
Mountain
Computer
INCORPORATED

MUSIC SYSTEM

OPERATING MANUAL

Mountain Computer tm
INCORPORATED

Manual Part Number 11-00227 -02

MusicSystem Operating Manual
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NOTICE

Install the enclosed component carrier at location B11 of the Apple II motherboard to correct any of the following:

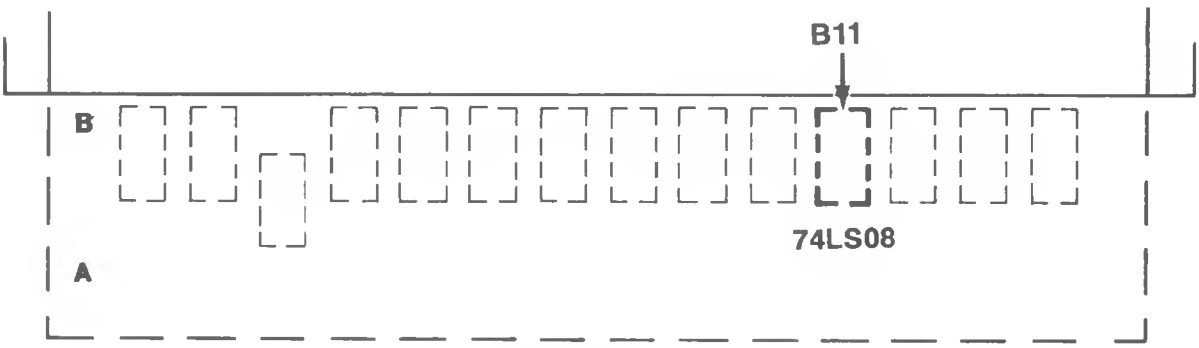
- Repeated interruption of MusicSystem output (middle of a song etc.) and computer hangs.
- Intermittent ability to boot the MusicSystem diskette.
- Problems loading or running programs when the MusicSystem cards are installed in the Apple.
- A disk drive boot sounds as if the drive is running slowly (drive makes strange noises etc.).
- MusicSystem is installed in the Apple but message

  ** ERROR ** MUSIC BOARD NOT IN MACHINE
  PRESS SPACE BAR WHEN READY

  is displayed.

INSTALLATION PROCEDURE

Turn the Apple II power switch to the OFF position and remove the 74LS08 at location B11 of the Apple motherboard as shown in Figure 1. Insert the 74LS08 IC in the enclosed component carrier and install it at position B11 as shown in Figure 2.



KEYBOARD
Figure 1

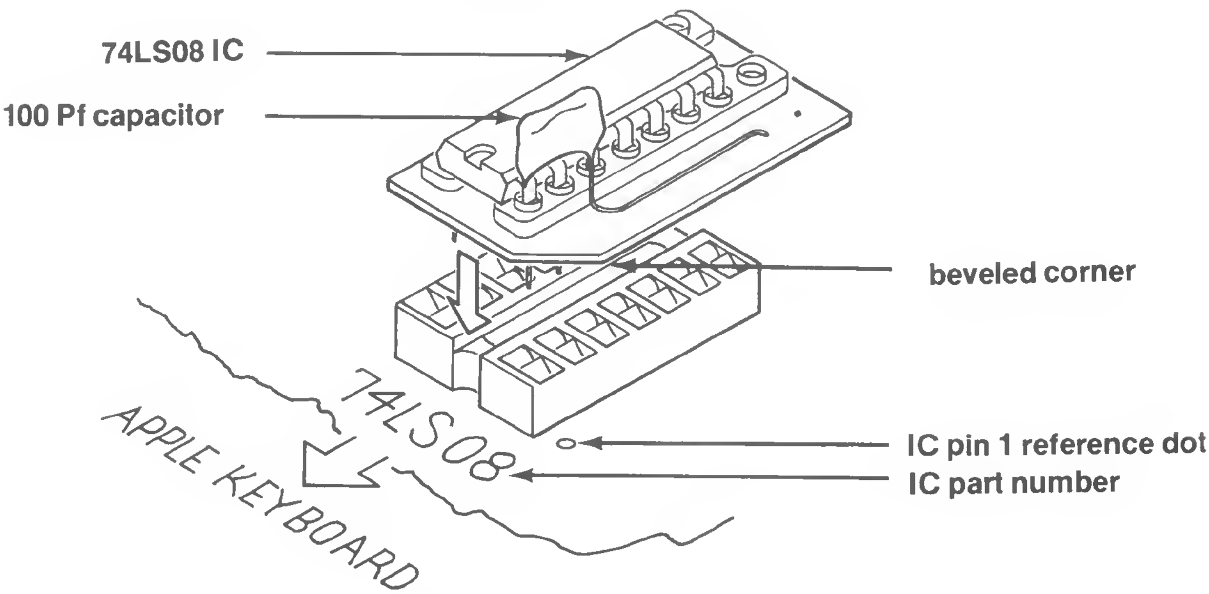


Figure 2

Chapter I Introduction

MusicSystem™ is an extremely powerful digital music synthesizer designed to run on any 48k Apple II* computer. Unlike "quasi-synthesizers", the MusicSystem utilizes the latest developments in additive synthesis. The hardware is supported by extensive and powerful software. MusicSystem's specially designed Music Editor allows you to compose musical compositions, save them on diskette, and print the musical scores on paper with a Silentype** printer. The Music Player plays the songs you create with the Music Editor, and the Instrument Definer allows you to create instruments using the principles of additive synthesis.

Hardware

The MusicSystem hardware has sixteen digital oscillators. Each of the sixteen oscillators generates a waveform that is completely programmable. These waveforms may be as simple as sine waves or very complex.

The MusicSystem hardware also includes a software-based volume control for each oscillator, and an overall volume control. The MusicSystem generates two output channels which connect directly to the auxiliary input on any stereo system through an audio output cable. If a stereo amplifier is not available you can attach a "Y" connector to the output cables to drive most high quality stereo headphones without an amplifier.

Included with the MusicSystem is a light pen which can be used for input and editing of musical scores in the Music Editor program.

Software

The MusicSystem software includes special programs which make the MusicSystem easy to use. The software allows you to

- compose and edit musical scores in standard musical notation.
- play your musical compositions or music we supply, with special options for changing the types of instruments and the locations of the stereo speakers.
- experiment and create the instrumental sounds used when playing compositions.
- arrange compositions with a variety of instruments.
- save your compositions on diskette.
- print musical scores.

Since the MusicSystem is a digital synthesizer, sounds unique to synthesizers as well as those of conventional instruments can be produced.

Traditional vs. Synthesized Music

Some people may wonder "Why use a computer to make music when we have a full heritage of acoustic and electronic musical instruments?" In answer to this question, the MusicSystem was not designed to replace more traditional instruments, but was created to provide an almost infinitely variable sound generating system coupled with music editing and composing capabilities, permitting you to explore the acoustics and structure of music for enlightenment and entertainment.

When programmed to the maximum of its capabilities it can perform in the manner of a multi-piece orchestra. Before each performance, the composer/programmer can change the arrangement of the composition in terms of the number and types of instruments to be used. Both the instruments we supply and instruments you have created can be used. The composer/programmer can also decide which parts will be played from which stereo speaker. These capabilities allow you to instantly change the character of a composition each time it is played.

The MusicSystem is easy to use and can be enjoyed by just about anyone. It is ideal for both the computer music buff and the musician with no computer music background. The MusicSystem can also form the basis of a musical education program.

Mountain Computer Inc. is committed to the continued development and support of the MusicSystem. In the future software enhancements and documentation updates will be made available. (Please send in your Warranty Registration Card so we will know where to send your MusicSystem updates!)

*Apple II is a trademark of Apple Computer Inc. of Cupertino, CA.

**Silentype is a trademark of Apple Computer Inc. of Cupertino, CA.

Chapter II Installation

This chapter provides the information you need to connect the MusicSystem to your Apple II computer and stereo system. It is assumed that

- you have used your Apple before and are somewhat familiar with its operation, and
- you know how to connect audio inputs to your stereo amplifier or receiver.

Hardware Requirements

Your Apple II computer must have the following features to use the MusicSystem:

- 48K RAM memory,
- one or more disk drives, and
- two or more adjacent unused peripheral slots (excluding slot #0) available for the MusicSystem hardware.
- In addition, if you wish to use the MusicSystem's printing capabilities, you need a Silentype printer.

Your stereo pre-amplifier or receiver must be capable of handling a standard tape input or auxiliary audio input. Monophonic systems will work, but you lose the capability for choosing which speaker you wish to use for each part in your composition. If you are using a monophonic amplifier, you will need to patch the two audio inputs together into a single input line.

By using a "Y" adapter cable to connect the dual RCA cables to a stereo headphone jack, the MusicSystem may be used without a stereo amplifier or receiver. The MusicSystem provides enough audio output to drive a pair of high quality headphones at normal listening levels. However, the quality of the sound is much improved when an amplifier or receiver is used.

Software Requirements

The MusicSystem will run on any standard 48K Apple II or Apple II Plus with DOS 3.3

Unpacking Instructions

The MusicSystem was carefully packaged in a specially designed carton to insure that it came to you in good working order. As you unpack each part observe how it was packaged and save all the packing materials. At some time you may have to re-package it yourself.

The MusicSystem boards should be handled with care. In particular, be careful not to cause any strain on the cables that connect the boards to the light pen, the audio output jack and the ribbon cable that connects the two Boards together.

While unpacking, take a few moments to fill out the MusicSystem Warranty Registration card. You will find a different serial number on **each** of the Music Boards. Be sure to enter **both** of these numbers on the Registration card before you return it to Mountain Computer.

The Registration card is our link to you and provides us with a way of sending you updates to the MusicSystem as they are developed. Sending in this card is your way of protecting your investment should any problem develop with your MusicSystem.

Parts List

Before assembling and installing the MusicSystem, check that all the parts listed here were included in your package.

2 Music System 5" floppy diskettes —
System 1 and System 2
1 MusicSystem Operating Manual
1 MusicSystem Product Evaluation Form
1 Warranty Registration Card
1 RCA stereo audio cable, 6 feet long

The following four parts are connected to each other and should be handled as one unit.

1 MusicSystem Board One
1 MusicSystem Board Two
1 light pen
1 audio output jack

If any part is missing, immediately contact the dealer from whom you purchased the MusicSystem.

Before Installation

Before you install the MusicSystem, be sure you have checked the following:

- The Apple's power is OFF! Never install ANYTHING in or remove ANYTHING from the computer when the power is on. Doing so can cause brain damage to your Apple or the MusicSystem.
- The stereo pre-amplifier or receiver is turned OFF.
- At least two consecutive peripheral slots are available for the MusicSystem Boards. (Slot #0 can't be used.)

Installation . . . Finally!

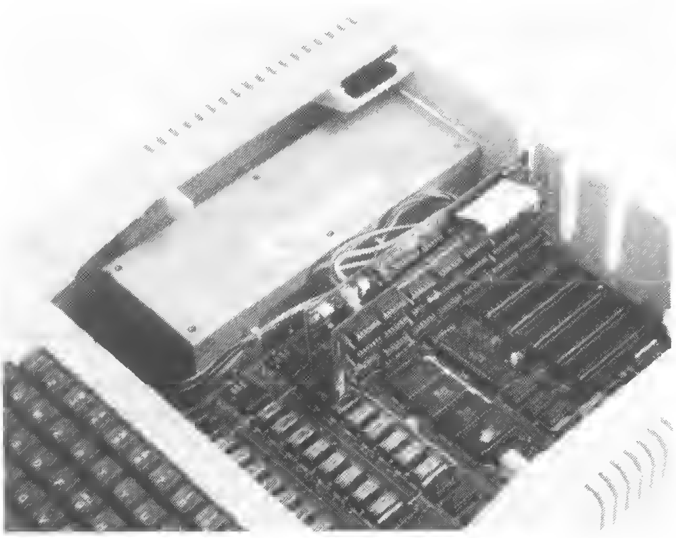
Thanks for bearing with us through these pre-installation notes. Although somewhat lengthy, they are designed to prevent possible problems **before** they arise.

If you made it this far, relax; the fun part is about to begin. Just follow these few simple steps and your MusicSystem will be ready to play music.

1. Turn OFF the power to the Apple and the stereo.
2. Remove the top cover of the Apple.
3. Touch the top of the power-supply unit (the silver or gold colored box near the left rear corner) to discharge any static electricity that may be lurking around you.
4. Arrange the ribbon cable that connects the two MusicSystem boards so that MusicSystem board I will be in the lower numbered slot, as shown in the illustration.
5. Carefully insert the MusicSystem boards in any two consecutively numbered slots (except slot #0).
6. Now thread the light pen cable out the back of the Apple case, bringing the light pen around to the front of the Apple. Put the light pen in a safe place while you finish installing the MusicSystem.
7. Carefully thread the audio output cable out the back of the Apple case.
8. Attach one end of the RCA audio cable to the dual plugs that you attached to the Apple's case in the last step. The other end of the RCA cable should be connected to the auxiliary input, tuner or tape monitor jack of the pre-amplifier or receiver.
9. If you are going to use the MusicSystem's print capabilities, install your Silentype printer with the method described in its accompanying manual. The printer can be installed in any slot except slot #0, but slot #1 is the standard.

   NOTE: The Silentype will not work with the MusicSystem if you have a High Speed Serial card plugged into your Apple.
10. Replace the Apple's cover.

That's all there is to installing the MusicSystem in your Apple. Now you are ready to begin. The next chapter, A First Tutorial, will start you on the way to becoming an expert with the MusicSystem.



Chapter III
A First Tutorial

This chapter discusses the uses of the Music Player and the Music Editor. In this chapter you will be led step-by-step through the procedures for entering musical compositions, saving the compositions on diskette, listening to them, and then printing them on paper. The Music Merger and the Instrument Definer will not be discussed here. The Music Merger is discussed in detail in Chapter VI, The Music Merger. An easy tutorial on using the Instrument Definer, and a detailed reference section on that subject appear later in the manual.

This section of the manual will not teach you every detail of using the Player and Editor, but it will start you on your way and give you the tools to write and play a wide variety of music. If, when you finish this chapter, you continue with the reference chapters that follow, you will be well on your way to becoming a MusicSystem expert.

Remember that the key to a successful learning phase is to keep your goals simple. Do not attempt to create a masterpiece on your first try. Instead, start with short and simple pieces. Then increase the complexity as you become familiar with the MusicSystem.

Copy Your Diskettes

Before you use the MusicSystem, in fact, before you even boot the MusicSystem diskettes, make backup copies of the MusicSystem software. Since diskettes are susceptible to wear and accidental abuse, you should NEVER use the MusicSystem Master diskettes that come with your MusicSystem except to make copies. Use the copies as your working diskettes, and store the Master diskettes away in a safe place in case your working copy is damaged, destroyed, or lost.

NOTE: A safe place for a diskette is somewhere away from extreme temperatures and magnetic fields (like those emanating from your video monitor). Store your diskettes vertically like phonograph records. (Diskettes like to sleep standing up.)

The MusicSystem diskettes are **double-sided**. Don't let this alarm you, even if you haven't ever used double-sided diskettes before. These diskettes work exactly like one-sided diskettes except that information is contained on two sides instead of one. We'll tell you how to make copies of both sides of the MusicSystem diskettes.

On both the MusicSystem diskettes the "front" sides (the sides with labels) contain the actual MusicSystem programs. The "back" sides (without labels) contain song or instrument files. The front sides of the diskettes are copied just like any normal single-sided diskettes using Apple's Copy program that comes with DOS. To copy the back sides, run the Copy program as you normally would, but insert the master diskettes **with the label facing down** as shown in the following illustration.

The blank diskettes you are copying onto should be inserted with the label side up, as usual.

Label the diskettes as you copy them. We suggest that you label the copy of the front side of the MusicSystem 1 diskette "System 1", and the copy of the back side of that diskette "Song Files". Then label the copy of the front side of the MusicSystem 2 diskette "System 2", and the copy of the back side of that diskette "Instrument Files".

NOTE: Make sure your new working diskettes are labeled clearly. If you insert the wrong diskette at the wrong time when you are running the MusicSystem, you may lose some of your work.

Booting For the First Time

Insert your working copy of the System 1 diskette in Drive 1, and turn the Apple's power on. If your Apple is equipped with the Autostart ROM, that's all you need to do. If not, proceed with the usual booting procedures.

The first thing that appears on your screen is the Main Menu. From this menu you may choose from three of the MusicSystem programs. You can also enter DOS commands at this point. The Main Menu looks like:

THE MUSICSYSTEM VERSION 2.0
DISKETTE—SYSTEM 1

COPYRIGHT 1980
MOUNTAIN COMPUTER INC.

1 MUSIC PLAYER

2 MUSIC EDITOR

3 MUSIC MERGER

CHOOSE FROM MENU
OR ENTER COMMAND

Each of the options on the menu represents a program in the MusicSystem. This chapter will concern itself only with these programs. The Instrument Definer is stored on the System 2 diskette. You will find detailed instructions on using the Instrument Definer in Chapter VIII.

Each program included in the MusicSystem has a different purpose, and combined they make up a powerful set of musical tools. The Music Player does just that, plays music. (It actually does quite a bit more than that, and its other functions will be discussed later.) The Music Editor lets you compose music (or enter music that already exists) and edit the compositions. The Music Merger is for merging small files that have been created with the Music Editor to create larger files.

Before you go further, turn on the power to your stereo amplifier or receiver. Always do this **after** booting the MusicSystem.

Listen to This! or The Music Player

Let's play some music! You can use the Music Player to play the symphony supplied with the MusicSystem. Select the Music Player from the Main Menu. You can do this in one of these three ways:

- Hold the light pen to the box surrounding the "1" next to "MUSIC PLAYER".
- Move the small cursor to the box surrounding the "1" by turning the knob on paddle 0, and press the paddle button.
- Type a 1 on the keyboard.

NOTE: If you are having trouble using the light pen, here are a few tips: If your light pen is connected properly and is in working order, the square you hold it against when choosing an option from the menu should flicker. If it does not flicker, the wires in the light pen may have come loose during shipment or installation and will have to be repaired. If the square does flicker, but the screen doesn't otherwise change, make sure you are holding the pen perpendicular to the screen when you you are choosing an option. If the room in which you are using the MusicSystem is brightly lit, especially if light is reflecting off the screen, you may have trouble too. Turning the brightness up on your video monitor will usually help.

When you have successfully chosen the Music Player from the list of options on the Main Menu, the following display will appear on your screen:

MOUNTAIN COMPUTER—
THE MUSIC PLAYER

ENTER COMP OR PLAY FILE NAME
(CTRL-C TO EXIT TO MAIN MENU)

NAME:

There are two kinds of files on the Songs diskette. The files that begin with the prefix PLAY. are PLAY files and can only be used with the Music Player. Files that begin with the prefix COMP. are Composition files (we'll call them Comp files). The Music Player converts Comp files to Play files. You can then save the Play files to avoid the converting time when you wish to play a song.

With One Disk Drive
If you are using one disk drive, remove the System 1 diskette and replace it with the copy you made of the Song Files diskette. Then type

MH SYMP

and press the RETURN key.

With Two Disk Drives
If you are using two disk drives, put your copy of the Song Files diskette in your second drive. Then type

MH SYMP, Sn, Dn

where n is the number of the slot in which the second drive is plugged and the drive number.

(Note that the MusicSystem uses the normal DOS conventions for specifying disk drives.) Then press the RETURN key.

Now the screen is reprinted and the song name, part numbers, part names, instrument names and speaker locations for this composition are displayed. Below this information is a set of the choices at your disposal for changing the composition. These choices include playing the composition, changing instruments or speaker location, setting all parts to mono, and saving the Play file. These choices are explained in full detail elsewhere in the manual.

For now you just want to play some music. Since we don't want to make any changes at this time press the RETURN key. The song called MH SYMP will begin to pour forth from your speakers. If, for some reason, you wish to stop the music before the song is finished, press CTRL-C.

When the song is finished (or aborted with CTRL-C) you'll get a prompt. If you have a dual drive system you are prompted to type the name of the next song to be played. If you have a single drive system you will be prompted to

INSERT SYSTEM DISKETTE INTO Sn, Dn
PRESS SPACE BAR WHEN READY

As before, n refers to the slot and drive numbers in which the disk drive is plugged. When you press the space bar you are prompted to type the name of the next song you wish to play.

Several pieces of music are provided with each MusicSystem in the form of Play files. From the Main Menu, you can do a CATALOG of the Song Files diskette to see a list of the available pieces.

To leave the Music Player and return the Main Menu to the screen, press CTRL-C.

With One Disk Drive
If you have a single-drive system, go to the Main Menu and insert the Song Files diskette in your disk drive. Then type

CATALOG

and press the RETURN key.

With Two Disk Drives
If you have a dual-drive system, go to the Main Menu and put the Song Files diskette in your second drive. Then type

CATALOG, Sn, Dn

As before, n represents the slot and drive numbers for your second disk drive.

The list that appears on your screen will look something like this:

PLAY.MH ROCK
PLAY.MH SYMP
COMP.FANTASIE1
COMP.FANTASIE2
COMP.FANTASIE
PLAY.FANTASIE

(The filenames on your diskette may differ.)

You can see that both Play files and Comp files are included on this diskette. Comp files will be discussed in detail later on in this chapter. For now you can play any of the files on the Song Files diskette with the method just discussed.

The chapter in this manual called "The Music Player" will tell you all about how to use the various Music Player options to arrange compositions creatively. We recommend that you finish this chapter before you turn to the Music Player chapter.

The Music Editor

The Music Editor is a tool for creating musical compositions. It was designed to approximate the process of composing a musical score on regular music staff paper, but with a few advantages. Instead of writing each note by hand on a tablet, the score is entered into the computer with the keyboard, light pen or game paddles.

In the rest of this chapter you will be instructed on how to enter the musical notation for the well-known song, America, from sheet music. Then you will be instructed on how to use the Music Player to play the song, and then go back to the Music Editor to make a few musical changes. If you follow along in this chapter, doing all the examples as they are presented, you will be able to do the same for any piece of music.

On the next page you will find the musical score for the song, America. You can copy from this score when entering the song in the Music Editor. You may find it convenient to temporarily remove the page so you won't have to flip pages back and forth as you are entering the composition.

From the Main Menu choose option 2, The Music Editor. In a moment, a musical staff and

America


My coun-try 'tis of thee, Sweet land of lib - er - ty
Of thee I sing; Land where my fath- ers died,
Land of the Pilgrim's pride, From ev - 'ry moun- tain side
Let free-dom ring.

the Music Editor's Main Commands Menu will appear on the screen. The Main Commands Menu appears as a high-resolution graphics display. To choose an option from this menu you will either point to a box with the light pen, or move the cursor to a square with the paddle knobs and press paddle button 1.

Choosing the Signature

Before actually entering the notes of the song you must specify the clef, the time, and the key for the composition. These comprise the Signature of the piece. Point to the square labeled "SIG CMDS" with the light pen. The Signature Commands Menu will appear on the screen.

NOTE: Throughout this chapter we will recommend the simplest technique for adding, changing, or deleting. Most often, this will be to use the light pen or paddle 0. If you wish to use alternate input and editing techniques, please refer to the Music Editor Reference chapter.

The song, America, has both a treble and a bass line. First let's enter the treble line. Since the treble clef is the default clef you don't have to choose the clef this time.

When entering the key for a piece, specify the number of sharps or flats in the key. Since America is in the key of G, which has 1 sharp, hold the light pen to the box marked "1#". The # symbol appears in the score immediately following the clef marker.

America is in three/four time so hold the light pen to the box marked "3/4". The 3/4 now appears in the score directly after the key signature. The composition signature is now complete so hold the light pen to the box marked "EXIT" to go back to the Main Commands Menu.

Entering Musical Notation

Now you are ready to enter the actual musical notes. Take a look at the sheet music for America. You will see that accents and dynamics are included. These can be entered in sequence with the notes, but for now don't bother with them; only concern yourself with notes, rests and measure bars the first time through. You can add the details later when you have more experience with the Music Editor.

A line of information is displayed near the bottom of the screen, below the menu. This is called the Status Line. It indicates the current status of some important aspects of the composition. The status information is broken up in the following way:

P=PART 1 ---- the part or voice you are working on (America has two).

M=1 ----------- the measure that is currently being worked on

O=4 ----------- the current octave. The current paddle octave displayed on the screen is the default octave. If you don't have paddles plugged into your Apple, octave 4 is the default. Octaves are discussed in detail in the section called "A Tutorial On Octaves".

M=NOTES --- the "mode" you are working in, Notes or Chords.

DUR=4 ------- the current duration for notes, chords, or rests. The default is a quarter note, which has a value of 4.

If you are using paddles, turn the knob on paddle 1 until octave 4 is displayed as the current octave. Now you are ready to enter the first note. To choose the note's pitch, rotate paddle 0 until the cursor is in the G position, as indicated by the sheet music for America1. Next, make the note's duration one quarter note by holding the light pen to the quarter note box on the Main Commands Menu. (Actually, since the duration is already set to a quarter note, as indicated by DUR=4 on the Status Line, you don't have to specify it. You only have to specify the duration when it changes. However, this time you need the practice.) Now press paddle button 0 to put the note in the score. Notice that the cursor has advanced automatically to the next position.

Continue to enter notes from the sheet music. Measure bars are indicated by pointing the light pen to the box that contains (guess what) a measure bar. Notice that when a measure bar is inserted in the score, the Status Line reflects the measure change.

When you reach measure 6 you will need to enter a rest that has the duration of a quarter note. Rests are easy. Change the duration to a

quarter note by pointing to the quarter note on the menu, just as if you were entering another note. Then press the light pen to the box with "REST" printed on it. The quarter note rest will appear in the score.

Enter the rest of the treble line in the same way. There are 14 measures in all. You will enter the bass line a little later.

Saving Compositions

There are four kinds of MusicSystem files that can be stored on diskette—Play files, Comp files, Idef (Instrument DEFinition) files, and Wave files. The Play and Comp files are stored on the Song Files diskette. The Idef and Wave files are stored on the Instrument Files diskette. (These files are created with the Instrument Definer and will be discussed in Chapter VIII.)

Play files are ready to play with the Music Player. Comp files can be edited with the Music Editor. Play files are made from completed Comp files.

The song you just entered now exists as a Comp file. Follow the directions below to save this file on diskette for later use. For now, save this file on the Song Files diskette. (You will probably want to eventually have many Song Files diskettes, perhaps cataloged by type of music.)

NOTE: You cannot save files onto either of the System diskettes. If you attempt this you will get the message

DISK FULL

If this happens to you, insert another diskette into the drive and again use one of the save procedures outlined here.

With One Disk Drive
If you are using one disk drive, while you are still in the Music Editor, type

SAVE AMERICA1

and press the RETURN key. (The 1 in the file name is just a version number so that, if you modify the composition, you can tell the versions apart.) The program will then reply

INSERT COMP FILE DISK IN Sn, Dn

PRESS SPACE BAR WHEN READY

Do as the screen instructs and insert the diskette that has your Comp files on it. Then press the space bar, and America1 will be saved on your diskette.

With Two Disk Drives
If you have a dual-drive system, put the diskette on which you wish to save the file in your second drive. Then, from the Music Editor, type

SAVE AMERICA1, Sn, Dn

where n is the number of the slot and drive numbers for the second drive. Then press the RETURN key.

When you have saved your composition, the Music Editor's Main Commands Menu will return to the screen. The file you just saved, America1, is still the current file being edited.

It is prudent to periodically save any file as you work on it, especially if it is a long file. If you save your file often you will be less likely to accidentally lose large quantities of work.

Leaving the Music Editor

To leave the Music Editor point the light pen to the box labeled "QUIT". You will be asked to verify your decision with

VERIFY (Y/N) →

Type a Y to leave the Editor, or type an N to cancel the QUIT command.

Caution! When you leave the Music Editor with the Quit command, **anything that Is in memory will be lost** (including the file you may have just spent all day working on). Make sure you have saved your file before you use the Quit command.

After you use the Quit command, the Main Menu will appear on the screen. Let's use the Music Player to listen to the newly entered song, America1.

We're Playing Your Song!

From the Main Menu select option 1, The Music Player. The new song, America1, should be on your Song Files diskette as a Comp file. You can choose this song from among the pieces on the diskette, just like you did when you played the symphony earlier in this chapter.

When it is loaded into the Player, the Comp file will be compiled into the internal form used by the Player. From the Player menu choose to play the Comp song, America1 and press RETURN. You will see the message

** WAIT--COMPILING **

When the file has been compiled, the Player menu will appear on the screen. Press the RETURN key to indicate that you want to play the song. You will be prompted to insert the Idef diskette that contains the instrument definitions. Insert this diskette and press the space bar. The instrument definition for the organ will then be "bound" to the internal file that contains the song score. The resulting file will become a Play file of the same name as the Comp file, and strains of America1 will begin to play. After the song is played, you can save the new file on your diskette. The new file will automatically be prefixed with PLAY. when you save it from the Music Player.

See the chapter called The Music Player for detailed information on using the Music Player.

Back to the Editor

After listening to your composition, AMERICA1, you may want to make some changes, or add the musical modifiers that serve to emphasize certain notes. You will do this in the following steps.

First, you must load the existing composition so that you can continue to work on it. With the Main Menu on the screen, select option 2, the Music Editor, from the menu or keyboard. Then with the Song Files diskette in the appropriate drive, depending on whether you are using one or two disk drives, type

LOAD AMERICA1

and press the RETURN key, or type

LOAD AMERICA1 Sn, Dn

and press the RETURN key.

When AMERICA1 is loaded, the score will appear on the screen along with the Main Commands Menu. Notice that the cursor is located at the beginning of the composition. If the notes that are in the first measure do not appear on the screen, you will need to enter a command that "re-paints" the screen. Change the duration, either with the menu or keyboard. The notes now appear.

Changing, removing and adding new musical information can all be described as "editing the composition". In this part of the tutorial, you will add some dynamics to the score for Part 1 (the treble part you already entered). Then, you will add another part (Part 2, a bass part) to the score so that the two parts can be played in unison.

Changing the Score

Compare the composition on the screen with the score in the sheet music. Suppose that you want to make a change to the duration of the first quarter note in measure 7. After all, you are the composer, and you do have rights to artistic revision! Perhaps you want to have the first note play a little longer, by adding a dot to it. Here's how:

Type GOTO 7 and press RETURN. The cursor is now located at the beginning of measure 7. Move the cursor one note to the right by holding the light pen to the " → " box on the Main Commands Menu. The pitch selection cursor should be located to the immediate right and parallel to the note to be changed.

Select the new, desired duration. In this case, it is a dotted quarter note. Then select CHNG DUR from the Main Commands Menu. The previous quarter note is transformed into a dotted quarter note.

After this addition of a dot to the quarter note, you will reduce another quarter note in duration. To change the final note in this measure from a quarter note to an eighth note, select an eighth note as the new duration and hold the light pen to the "CHNG DUR" box on the Main Commands Menu. Of course, the cursor needs to be located to the immediate right and parallel to the note that is to be changed.

Adding Dynamics

You may have noticed while listening to the composition that no emphasis was placed on those portions of the score that "normally" get an extra bit of emphasis. The composition may have sounded unnaturally flat or "machine-like". During this part of the tutorial, you will add dynamics to several sections of the score.

To add the mezzo forte (mf) in measure 1, first type "GOTO 1". Then move the cursor to the immediate right of the 3/4 time signature by using the " → " symbol on the Main Commands Menu.

Dynamics are inserted just like notes. To insert the mezzo forte at the beginning of the score and the forte in measure 7, choose the Sound Commands Menu ("SOUND CMDS") from the Main Commands Menu. With the cursor located in the proper place horizontally via the " → " box, choose the desired dynamic (mf or f).

Speaker Location
You can designate what parts of the composition will come from which speaker, or the "spatial location" for each part. Type GOTO 1 to move the cursor to the beginning of the composition, and once again, re-locate the cursor to the immediate right of the "mf" dynamic you just entered.

From the Main Commands Menu, choose "SOUND CMDS". When that menu appears, you can define how the sound should be arranged. Hold the light pen to the box marked "LEFT" for all sound to come from the left speaker, "RIGHT" for all sound to come from the right, or "BOTH" for an equal amount from both channels.

For now, select RIGHT to place this part in the right channel. An asterisk appears in the score, and the status line displays the message:

SPATIAL LOCATION = 9

This means all sound of this part will come from the right speaker.

Instrument Assignment
Once you have finished editing the composition, you can assign an instrument to play this part of the composition. You can choose from the instruments on the Instrument Files diskette. This instrument will be assigned to PART 1 of the composition. For now, move the cursor to the beginning of the score, and type

INST BRASS

This assigns a brass-like instrument to the composition. A star appears at the beginning of the score and the instrument name appears on the status line. When you have completed editing the composition for the first part, save the composition again to protect the work that you have done so far. With the Song Files diskette in the appropriate drive, depending on whether you are using one or two disk drives, type

SAVE AMERICA2

If you wish, you can leave the Music Editor and listen to AMERICA2 from the Music Player, or you can continue with the tutorial, and add the second part. If you leave the Editor and then return, be sure you are using the most recent version of the composition by typing

LOAD AMERICA2

Each time you begin to edit an existing file, the Music Editor automatically selects the first created part, Part 1, as the current part and places the cursor at the beginning of that part.

Adding Another Part
Now you are going to add a second part to the composition. The procedures for beginning another part are simple. Type

ADDP PART 2

In this case, you will call the second part PART 2. You did not specify a part name the first time you used the Music Editor because the first part is automatically assigned for each new composition.

Part 2 is a bass part, so use the bass clef. The procedures for entering the second part are identical to the procedures for the first part. Follow these abbreviated instructions:

Clef: "BASS" clef from the SIG CMDS menu.

Key: "1#" from the SIG CMDS menu.

Time: "3/4" from the SIG CMDS menu.

Notes: Enter all notes, rests and measure bars for PART 2, following the sheet music supplied on page III-4, until you reach measure 4.

Accidentals: In measure 4 (and other measures), various accidentals are used in the score for Part 2. To enter individual accidentals, **the sharp, flat or natural must be entered immediately after entering the note it is affecting**. With the cursor to the immediate right of the note to be "accidentalized" (i.e., the note just entered), choose the desired flat or sharp.

See the discussion of accidentals in the Music Editor Reference chapter for further details.

Spatial Location (or Speaker Location) for Part 2: Spatial location is set in the same way as for Part 1. In general, a distributed arrangement of sound produces better results than attempting to place all parts within the same channel.

For this reason, choose LEFT from the SOUND CMDS menu for Part 2 of this composition (Part 1 was RIGHT). The cursor should be at the beginning of the part when specifying the Spatial Location.

Instrument Assignment for Part 2: Since you used a brass instrument for Part 1 (which contains the treble portions of the composition), use an organ sound for Part 2 (which contains

the bass clef portions of the composition). After moving the cursor to the beginning of the part, type

INST ORGAN

This completes the entry of Part 2. Remember that you began this editing session by loading AMERICA2. When you quit the editor this time, save the composition with the name AMERICA3. AMERICA3 will contain all the work entered for Part 1, plus the work just entered for Part 2. To save the entire new composition, type

SAVE AMERICA3

Quit the Music Editor, and go to the Music Player to hear the results of this final editing session.

If you have a Silentype printer plugged into your Apple and you want to print a copy of the Musical Score for the song, America, continue with the next section. If you don't have a Silentype printer, skip to the section called "The Composition Library".

Printed Music

The MusicSystem lets you print the musical scores that have been entered with the Music Editor. The scores are printed vertically on the paper in one long strip. Entire compositions or just specified measures can be printed. Likewise, all the parts in the composition, or one or more specified parts can be printed. In addition, musical scores can be printed in either "double" or "single" size.

The printing of musical scores is done from the Music Editor. Choose the Music Editor from the Main Menu. Then load your latest version of the song, America, from the Song Files diskette.

NOTE: It is VERY important to save your work before you print. It usually takes the Silentype a few minutes to print a musical score. If the power was to go off during that time, you would lose whatever was in memory, namely, your most recent song.

The Print command is a non-menu command. That is, it doesn't appear on a menu. To invoke the Print command, type

PRINT

but don't press the RETURN key yet. Instead, press the space bar once. At this point you have several options. This section will lead you step by step through all of them. Then, when all the options have been covered, you can execute the Print command to get a printed copy of America3. If you leave an option out, the default value for that option will be assumed.

First use the Part option to indicate that you want to print the treble part, or Part 1, of America3. Type

PT 1,

(Make sure there is a space after PRINT.) You can print a range of parts at one time if you wish. If you wanted to print, for example, parts 1 through 3 in a four-part piece, you would type

PT 1/3,

Parts 1 through 3 would then be printed sequentially, first Part 1 in its entirety, then Part 2, then Part 3. Don't forget to put in the comma. If you forget it, you will get a Syntax Error when the Print command is executed.

If you don't type in a Parts option, all the parts in the composition will be printed.

Next let's use the Measure option. For now you want to print only measures 1 through 6. To indicate this type

M 1/6,

Again, don't forget the comma. If you wanted to print only one measure you would type an M followed by that number. If you don't specify a measure or measures, all measures in the specified part or parts will be printed.

The slot option is for specifying which slot your printer is plugged into. If your printer is plugged into slot #4, for example, you would type

S 4,

If your printer is plugged into slot #1, you don't have to type the slot option, as 1 is the default.

NOTE: Make sure you specify the right slot number for your printer. If you give the wrong number, the program will "hang" and you will probably lose anything that is in memory.

The Octave option allows you to print scores in different octaves than the one in which the score appears on the screen. As you become familiar with the MusicSystem you will find that it is sometimes convenient to print a piece in the "wrong" Octave. To print a score in, for example, Octave 3, type

O 3,

Octaves can be a little tricky. See the section called "A Tutorial on Octaves" for more information on them. For now don't specify an octave. This will cause the default octave, the one in which the piece is displayed on the screen, to be printed.

If you want to print the score in the single size don't type anything else, as the single size is the default. If you want to print in the double size, type

DB

These five options can be entered in any order. They must be separated by commas. The last option does not have to have a comma after it.

When all the options are as you want them, press the RETURN key, and the score will begin to print.

You will find that the Silentype will occasionally print slightly out of line. Don't be alarmed. This is normal for the Silentype.

The Composition Library

Put your Song Files diskette in your disk drive (drive 2 if you are using a dual drive system). The CATALOG command from the Main Menu shows you the contents of the diskette. These files comprise your composition library. All commands for file manipulation apply to these composition files. If you wish, you can DELETE the AMERICA 1 and AMERICA 2 compositions since they are the earlier versions of AMERICA3. From the Main Menu use the standard DOS command, for example,

DELETE COMP.AMERICA1

It is wise to backup your MusicSystem files to another diskette. The DOS COPY program can be used to backup an entire diskette, or the Music Merger can be used to backup individual files. See Chapter VI, The Music Merger, for information on using the Merger to copy files.

A Tutorial on Octaves

Because the display and control of Octaves in the Music Editor is somewhat complex and is, in fact, constrained by the size of the display area on the screen, a special tutorial has been developed to provide extra assistance in learning to control the use of Octaves. This tutorial is **separate and apart** from the preceding tutorial on using the MusicSystem.

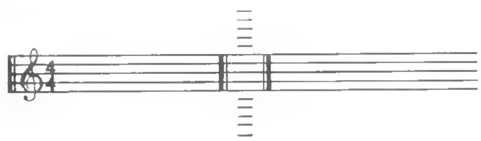
For additional information on Octaves, please see the section called "A Note About Octaves" and the discussion of the Octave command in the Music Editor Reference Chapter of this manual.

Octave Display Conventions

The purpose of this tutorial is to provide you with an intensified display of the way the Octave commands function in the Music Editor. This tutorial is structured to provide maximum visual orientation to the way the octaves are displayed on the screen in different situations.

Remove the game paddles from the Apple and boot the MusicSystem as you normally do. As always, be sure you are using a copy of the original disk, not the master.

Select the Music Editor from the Main Menu. Since the game paddles are not installed, you will need to use either the light pen or the keyboard. The first display looks like the following illustration.



Notice that there is no pitch select cursor (the left-pointing arrow) on the screen; without the paddles it serves no purpose and therefore is not displayed. Likewise, the octave number that normally appears below the staff lines is not present; again, this is because the paddles are not present. That octave display is a number associated with the octave determined by the position of paddle 0. Finally, note that the Status Line displays the default octave value of 4 (O = 4).

Change the octave to octave 0, by typing

:0 (CR)

The only change is on the Status Line, now O = 0. Next, enter the first two notes of the C scale. Type

C D (CR)

These notes appear on the screen. Enter a measure bar; type

M (CR)

The next step is to raise the octave, and enter the same notes in octave 1. First, type

:1 (CR)

Notice that the first two notes shift to a lower place on the screen. Now add notes C and D in octave 1, by typing

C D (CR)

Finally, add a measure bar by typing

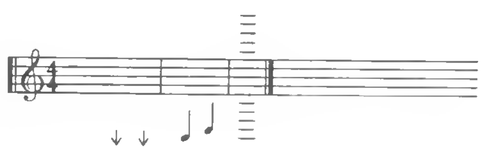
M (CR)

Whenever the Editor processes a command that changes the current octave value, the screen is re-painted and all existing notes are shifted from their previous display location.

Let's repeat the process, changing octaves and adding notes as follows. Type

:2 (CR)

As soon as this is processed, the original two notes in measure 1 are replaced by two downward-pointing arrows. This indicates that the notes are in an octave **outside the display-able range on the screen**. At this point, the screen should look like this:



Now add the next two notes for measure 3, again C and D. Type

C D (CR) followed by

M (CR)

The two downward-pointing arrows are now scrolled off the screen display.

Change the octave to 3, by typing

:3 (CR)

And add two more notes, C and D and another measure bar:

C D (CR) followed by M (CR)

Continue this process, until you have entered notes C and D in octave 7. Then add the note B in octave 7 and a final measure bar for this measure; this is the highest note possible in the treble clef.

You have created a "sampler" of the entire range of octaves for the treble clef.

Repeat this exercise for the bass clef, and then the system clef. Begin the next measure (measure 9) by typing

BASS (CR)

and enter, in sequence

Octave 0, Notes C and D, Measure Bar
Octave 1, Notes C and D, Measure Bar
Octave 2, Notes C and D, Measure Bar
Octave 3, Notes C and D, Measure Bar
Octave 4, Notes C and D, Measure Bar
Octave 5, Notes C and D, Measure Bar
Octave 6, Notes C and D, Measure Bar
Octave 7, Notes C, D and B, Measure Bar

This is the full range "sampler" of the Bass clef. Now type

SYSTEM (CR)

and enter in sequence

Octave 0, Notes C and D, Measure Bar
Octave 1, Notes C and D, Measure Bar
Octave 2, Notes C and D, Measure Bar
Octave 3, Notes C and D, Measure Bar
Octave 4, Notes C and D, Measure Bar
Octave 5, Notes C and D, Measure Bar
Octave 6, Notes C and D, Measure Bar
Octave 7, Notes C, D and B, Measure Bar

This the full range "sampler" of the System clef.

SAVE this complete Comp file to the name OCTAVETEST. When the file is SAVEd, go to the beginning of the file by typing

GOTO 0

Now, return to octave 0 by typing

:0 (CR)

A series of ascending notes are displayed. Use the SCROLL RIGHT command (Main Commands Menu) or CTRL-A from the keyboard to scroll through the score.

Notes disappear and are replaced by upwards pointing arrows until the bass clef symbol is passed. At that point, you are back in octave 0, and the notes are again displayed. As you scroll beyond the displayable range of notes in the bass clef, the notes are again replaced by upwards pointing arrows.

As you enter the measure with the system clef, notes are again displayed. As before, when the octave increases, the notes are displayed as upward pointing arrows.

Return to the beginning of the composition, again by typing

GOTO 0

and set the display of octaves to the higher registers by typing

:7 (CR)

Scroll through the composition as before, and observe the effect of having notes on the screen in an octave lower than the octave set by the keyboard octave command. This time, you see a series of downward pointing arrows until you reach a set of notes in the current displayable range.

This ends the first portion of the Octave Tutorial. At this point, you should be sure you have saved OCTAVETEST. Now turn off the Apple, and re-install the game paddles. Re-boot the system, select the Music Editor and load OCTAVETEST.

With the paddles installed, notice the left-pointing pitch select cursor and the number displayed below the staff lines. The first notes in the composition are displayed as downward pointing arrows. Scrolling through the composition, note that **only those notes in the range of the octave number displayed below the staff lines** are displayed as notes; all others are displayed as upward or downward pointing arrows. The octave range controlled by paddle 0 is four octaves.

When the game paddles are plugged in, they control the display of octaves. **However, the display on the screen is not immediately affected by rotating paddle 0. The screen is only re-painted by processing certain commands.** You can re-paint the screen by changing the duration, for example. Doing this causes no change in the composition, but re-paints the screen according to the current value of paddle 0.

For example, move to measure 2, by typing

GOTO 2

This measure contains notes C and D in octave 1. Turn paddle 0 all the way **counterclockwise**. When you type

Q (CR)

the screen is re-painted, and all the arrows are re-painted as notes. Now rotate paddle 0 all the way **clockwise**. When you type

Q (CR)

the screen is once again re-painted, and all the notes are replaced with arrows.

Experiment with paddle 0, the keyboard octave command, and the relationship of the two until you feel that you understand the way that octaves are displayed and controlled. The more you use a wide variety of clefs, and a wide range of octaves **in simple, experimental compositions**, the more you will understand this somewhat complex part of the Music-System.

Three general rules apply to the display and control of octaves:

1. The **bass** octave is always in the key of C.
2. The C note in the bass octave will be displayed as middle C.
3. When the paddles are plugged in, the octave control via the paddles have priority over the keyboard octave in regard to display of octaves. Changing the octave with the paddle does not automatically re-paint the screen.

Chapter IV The Music Editor, a Guide to Use

This chapter is a guide for the novice user of the MusicSystem. It provides information about capabilities and procedures rather than technical details, and will assist you in understanding the Music Editor. For a concise explanation of each Music Editor command, see the chapter entitled "The Music Editor, Reference".

The first section in this chapter gives a general overview of the Music Editor's capabilities. The second section is concerned with disk I/O (saving and loading files, etc.) and discusses musical compositions as disk files. The third section, called "Within the Editor", discusses the actual manipulation of composition files and musical "events" from within the Music Editor. (A musical event is any change made to a composition file with the Music Editor.) Lastly, the Music Editor's printing capability is discussed.

How It Works — An Overview

The Music Editor is a tool for creating and modifying musical composition files. The four main functions of the Music Editor are

1) Input and editing of data
2) Display of musical scores
3) Printing musical scores
4) Loading and saving compositions

Input and Editing of Data
To make entering musical notation as simple as possible, three separate input techniques have been developed. These are the light pen, game paddles, and keyboard commands. Not all commands can be used with each input device, but the keyboard can be used for anything except moving from one menu to another.

Display of Musical Scores
The special graphics display capability provides realistic looking musical scores. A small number of musical notation symbols are represented by the appearance of stars (or asterisks) in the score. The meaning of a star is displayed on the status line when the cursor is located to the right of the star.

Printing Musical Scores
A replica of the screen display can be printed on paper with a Silentype printer. Entire musical pieces or portions of them can be printed in two sizes.

Loading and Saving Compositions
Compositions in the form of Comp files can be loaded from and saved to diskettes from within the Music Editor. Only Comp files can be used with the Music Editor.

Compositions as Disk Files

This section discusses the ways in which compositions are manipulated on diskettes as disk files.

Naming Compositions
Compositions in the Music Editor may have any name as long as these conventions are observed:

1. Composition filenames may include any alpha or numeric characters, spaces, and special characters except commas and control characters.
2. Composition filenames may be from 1 to 25 characters.
3. Composition filenames will automatically be appended with the prefix, "COMP." when entered in the disk catalog from the Music Editor.

There are two ways to name compositions:

1. Compositions can be named with the SAVE command from within the Editor.
2. An existing composition can be renamed from the Main Menu with the RENAME command (see the Apple DOS manual for details). If you rename a file, you must include the "COMP." prefix.

Loading a Composition
You can load Comp files from diskette when the Editor's Main Commands Menu is displayed on the screen. Type

    LOAD

followed by the name of the song you wish to load, and then press the RETURN key. Don't type the COMP. prefix.

Saving a Composition
SAVE should be used frequently during an editing session to protect the work you have just completed. SAVE the contents of the file from within the Editor. Two types of saves may be used:

1. You can SAVE a composition under an existing filename. If you SAVE a file to a diskette that already has a file of that name, you are prompted

   FILE EXISTS AND WILL BE
   OVERWRITTEN

   VERIFY (Y/N) →

   Press Y to save the file and overwrite the existing file of the same name. Press N to cancel the SAVE.
2. You can SAVE a composition under a new filename. Type SAVE "new filename" and press the RETURN key. The file will be saved under the new name you specified prefixed with the COMP. prefix.

All the usual DOS conventions for specifying slot and drive numbers are used in the Music Editor. See the Apple DOS manual for information on specifying slot and drive numbers.

If you are using a single drive system, when you attempt to SAVE a file to the same slot and drive location that contains the system diskette, you are prompted

INSERT COMP FILE DISK IN Sn, Dn

PRESS SPACE BAR WHEN READY.

Remove the system diskette, place the Songs diskette in the same drive, and press the space bar. The file will be saved, and you will see

INSERT SYSTEM DISK IN Sn, Dn

PRESS SPACE BAR WHEN READY.

Replace the system diskette, press the space bar, and you are returned to the file you were just editing.

Although Comp files appear to be binary files when you look at the diskette catalog, they are actually a type of file that is unique to the MusicSystem. Comp files cannot be BLOADed or BSAVEd. See the chapter called The Music Merger for information on copying Comp files.

Deleting a Composition

Deleting a composition is done from the Main Menu using the DOS DELETE command. The syntax for the DELETE command is the same as for regular Apple DOS; i.e., you must include the full filename, including the PLAY. or COMP. prefix. For example,

DELETE COMP.AMERICA2

would cause the file called COMP.AMERICA2 to be deleted from the diskette.

Within the Editor

The commands discussed in this section are actually done from within the Music Editor. These are the commands you will use to write and edit compositions.

Beginning a New Composition

You can start a new composition in one of two ways:

1. Use the Music Editor NEW command to erase the contents of the composition file currently in memory.
2. Enter the Music Editor. Then enter musical data without loading a Comp file.

Editing an Existing Composition

To edit an existing composition, enter the Music Editor and load the file you wish to edit. If you are using a two drive system, specify slot and/or drive numbers. Standard Apple DOS conventions for slot and drive numbers apply. If you are using a single drive system, follow the screen prompts as usual.

When the composition is loaded, the musical notation will appear on the screen. You can then begin to edit.

Scanning Through a Composition

You can search for a particular item from within a composition either by moving within a part, or from part to part.

Movement within a part is controlled by the Left and Right cursors located on the Main Commands Menu. The keyboard command for left movement is CTRL-Q. The keyboard command for right movement is CTRL-A. To move to a specific measure, use the GOTO command. See the Reference chapter for details.

Movement from part to part is controlled by the Up and Down Scroll cursor commands from the Main Commands Menu. These are symbolized by up and down pointing arrows. The keyboard command for "move to previous part" is CTRL-W. For "move to next part" the command is CTRL-S.

Music Event Manipulation

NOTE: Throughout this manual, you will notice the term "musical event". A musical event is any structure of musical notation including notes, chords, rests, bars, and **all other** elements of a musical score. Notation marks and

symbols are musical events and all special features of the Editor that are not found in conventional music are musical events.

Inserting Musical Events

The Music Editor is in insert mode by default. When an event is to be included in a composition, it is selected (by keyboard or menu) and entered (by carriage return or button 0). To perform editing functions other than insertion (i.e., deleting, changing), select the function from the menu or keyboard. See the Reference chapter for a list of all functions performed by the Music Editor.

Deleting Musical Events

The Music Editor allows for deletion of musical events. Place the cursor to the immediate left or right of any event and use the DEL RIGHT or DEL LEFT commands. These commands are found on the Main Commands Menu. From the keyboard, the DEL RIGHT command is →, the DEL LEFT is ←. Using the REPEAT key with these commands from the keyboard causes successive deletion of musical events. Be careful when using the REPEAT key as it deletes events repeatedly, and you may accidentally delete something you didn't want to.

Changing Musical Events

Two special commands allow changes to notes and chords without specific delete and re-insertion. These are Change Pitch and Change Duration. To change the pitch of a note within a chord, position the cursor to the right of the note and select CHNG PITCH. CHNG DUR is used in the same manner to change the duration of a note or chord. The Change Pitch command from the keyboard is CP, the command for Change Duration is CD.

Accents, dynamics and other modifiers to notes and chords are used as described in the Reference chapter to change the way a note is played.

Printing Musical Scores

You can print all or part of a musical score if you have a Silentype printer installed in your Apple. The printed score is a copy of the display that appears on your video screen. To print a composition, follow these steps: While in the Music Editor save the composition you wish to print. Then type the Print command with the options you want, and press the RETURN key. For specifics on using the Print options, see the section on printing in Chapter V Music Editor, Reference.

Chapter V
The Music Editor, a Reference

This chapter provides detailed information about the Music Editor. It is intended to be used as a reference. If you need basic instructions in using the Music Editor, please see the chapters titled "A First Tutorial" and "The Music Editor, A Guide to Use".

The Screen Displays

The Music Editor has three types of screen displays. Each display has a different function.

The Main Menu

The Main Menu is the first thing you see when you boot the System 1 diskette. While not technically a part of the Music Editor, it is the means of gaining access to the Music Editor.

The Upper Screen

The Upper Screen is a high resolution screen that displays the score that is being edited. During most Music Editor operations, only one set of staff lines is displayed. However, when "System Clef" is used in a composition, both the treble and bass staff lines are displayed. The small "backwards arrow" cursor on the Upper screen is used to indicate current beat position and to control vertical position within the score.

The Lower Screen

The Lower Screen is a high resolution screen that displays one of a set of menus, a composition status line and a keyboard input line. Two cursors are used on the lower screen; one to select items from the menus, the other to manage keyboard input.

The Status Line

The Status Line displays the following information about the composition and the position of the cursor within the composition. From left to right on the status line are

1. PART — the current composition part is identified by a user supplied alphanumeric name. When the cursor is moved from part to part, the name displayed on the status line will change.
2. MEASURE — the current measure is represented by a number. The number of the first measure in a composition is 1.
3. OCTAVE — the current octave value is represented by a number from 0 to 7; 0 is the lowest octave and 7 is the highest.
4. MODE — either NOTES or CHORDS can be used. The current mode definition is displayed on the status line.
5. DURATION — the current note duration designation is displayed. Duration remains constant until a different duration is selected. (See the section called "Duration".)

The Status Line is supplemented by a message when the Music Editor must inform you of a special condition in the score at the current cursor location. These conditions are

A. STARRED ITEMS — when the cursor reaches a "star", its definition is displayed on the status line. Starred items include: instrument assignments, synchronization marks, spatial location marks, and tempo changes. Additionally, when the cursor is located at a key signature of C (no sharps or flats), a star appears and the key value is displayed. This is because the Key of C does not require a notation in the score. When the cursor advances beyond the star, the normal status line reappears.

B. ERROR MESSAGES — when an input error is detected or illegal combinations of musical events are attempted, an error message appears on the status line and the cursor is placed on the error. The error message remains on the screen until the next command is executed. The error messages are listed in Appendix C in the back of this manual.

The Keyboard Input Line

The Keyboard Input Line is below the status line, and is used for entering commands via the keyboard. Most musical events can be input or edited via the light pen and game paddles, but some require the keyboard. A keyboard command is entered by typing the command and then pressing RETURN. No undefined control characters may be entered on the keyboard input line.

The Cursors

Keyboard Command Cursor: The keyboard command cursor is located on the Keyboard Input line. The following chart shows the available cursor control commands.

COMMAND	DEFINITION	FUNCTION
CR	carriage return	execute line from beginning to location where cursor is -- truncate rest of line
CTRL-E	execute	execute entire line
— or CTRL-H	backspace	move cursor to left, invert character under cursor
-- or CTRL-U	forward	move cursor to right, invert character under cursor
CTRL-D	delete char.	delete character under cursor
CTRL-X	delete line	delete entire line, move cursor to beginning of line
CTRL-I	insert	insert characters at cursor, line to right. Insert is turned off by any of the above commands except itself.

Illegal symbols, such as "?" or integers greater than 255, will not be accepted by the Editor. If you try to enter an illegal symbol the cursor will be moved to the offending character and wait for you to type an acceptable one.

Editing Cursor
Paddle 0 controls the vertical position of the cursor within the staff. By turning the paddle, the small cursor will move up or down the staff lines.

Paddle 0 also changes the octave of a **single note**. By turning paddle 0, the octave for that note will increase or decrease beyond the range of the current octave. When a different octave is entered, a number associated with that octave is displayed below the cursor in the middle of the screen. The number of the octave on the status line is only associated with the keyboard pitch entry. The number of the octave displayed on the top of the screen is associated with paddle 0.

NOTE: These two representations of octave have no relation to each other.

A special discussion of the way that octaves are displayed and controlled in the Music Editor is included in this chapter under the heading "A Note About Octaves". Please refer to that section for full details on octaves.

The editing cursor also indicates the horizontal location in the score. That is, the sequential location in the composition. Current score position is not controllable from paddle 0, but from the Left Cursor or Right Cursor selections on the Main Commands Menu.

Menu Selection Cursor
Paddle 1 controls horizontal motion from box to box in the menu. By turning the paddle, the menu selection cursor advances left or right. When the cursor is located within a box, you may choose that box by pressing button 1 (located on paddle 1). If the cursor is moved beyond the left-most box on a line, it moves up a line and reappears in the right-most box. Similarly, if it is moved beyond the right-most box, the cursor moves down and reappears in the left-most box. Paddle 1 with button 1 is an alternative to using the light pen for menu selection.

Input and Editing Techniques

The techniques for entering musical events were designed for flexibility. You can choose the light pen, game paddles or keyboard according to your own preference. However, some input and editing operations require a specific technique. The musical events requiring special input techniques are described in this chapter. Experiment with each technique to find the style that is easiest for you.

Insert is the default input mode of the Music Editor. Unless a command modifier is included with a command (regardless of technique), that musical event will be inserted. Any command other than an insert (such as a delete or change) needs to be specifically designated.

Keyboard Input
Any command may be entered via the keyboard, except moving from one menu to another. Type the command followed by the appropriate objects and modifiers. Syntax for each command is described in this chapter. In addition, you will find a list of keyboard commands in Appendix B.

Light Pen Input
The light pen can select a musical event which appears on one of the menus in the Music Editor. The event chosen will affect the composition at the current cursor location. The procedure is to first move the cursor to the event to be affected, then hold the light pen to the box containing the event to be inserted. The boxes preceding the selected box will "blink" in sequence up to the selected box. The selected box will "blink" rapidly while the Music Editor is verifying the choice of this box. HOLD THE LIGHT PEN TO THE BOX UNTIL THE BLINKING STOPS. This insures that the choice is accurately made. To repeat a selection with the light pen, continue holding the pen to the screen until the musical event has been generated the proper number of times.

The light pen cannot be used to position an input. This must be done with paddle 0 (for vertical location within a part). The light pen may select Cursor Forward, Backward, and Scroll Parts Up and Down. The pen only selects.

Paddle Input
Paddle 1 is used to select menu items from the Main Commands Menu in much the same way as the light pen. Turning the paddle moves the menu selection cursor to any box in a menu. Pressing the paddle button selects the musical event.

Paddle 0 has two main functions. Rotating the paddle knob causes the cursor to move up and down in the score. Pressing the paddle button inserts the musical event you have chosen with Paddle 1. Paddle 0 is also used to change the display octave.

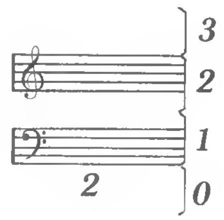
When using the paddles to select or insert a musical event, be sure to press the buttons **firmly** to insure that the selection or insertion is made.

A Note About Octaves

The way that octaves are displayed in the Music Editor is fairly complex and needs elaboration.

The "scrolling score" portion of the screen can display up to 4 octaves at a time. Since there are eight functional octaves (0 - 7) for any part in a composition, all octaves are available, but cannot be displayed on the screen at one time.

On the screen **between the menus and the score** appears a number which indicates the value of the octave in the middle of the screen portion. For example, if the number 2 appears, the range of octaves displayed is as follows:



Thus, any notes entered in octaves 0 through 3 appears on the screen. Specifically, this means that the middle position on the screen is middle C in octave 2.

Notes Outside the Displayed Octave Range
If a note that has been entered is above or below the currently displayed range of octaves, that note is indicated by the appearance of upward or downward pointing arrows **directly above or below the horizontal location of the note in the score.**

Changing the Displayed Octave Range
If you have paddles plugged into your Apple, you may change the range of octaves displayed on the screen by rotating game paddle 0. As the paddle is turned, the octave value (displayed in the middle of the screen) changes. The octave value on the Status Line does not change when you change the range of octaves with the paddles. Notice that the display of the staff lines and the octaves does not change.

If the paddles are not plugged into your Apple, you can change the octave range from the keyboard by typing a colon (:) followed by the octave number you wish and then pressing RETURN. When you change the octave range with this method, the octave value that appears on the Status Line will change accordingly.

To re-display the current range of octaves, you must enter a command. A simple command to use when you want a re-display of the screen is to re-select the current duration. After doing so, the screen is "re-painted" with the appropriate octaves displayed.

Entering Notes; Keyboard vs. Paddle 0

The procedures for entering notes in a specific octave range are different depending on whether you are using the keyboard or Paddle 0.

When entering notes from the keyboard, the note will be located in the octave displayed on the Status Line on the bottom of the screen. When entering notes with the game paddle, the note will be located in the octave displayed on the "scrolling score" portion of the screen.

To change the value of the octave associated with the keyboard, type

:n (RETURN)

where n is the value of the desired octave (0-7). If you give a value that is out of range (less than 0 or greater than 7), you will get an error message.

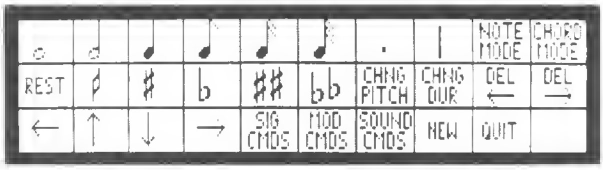
This capability for changing the octave allows you to enter notes of different octaves even if the paddles are not plugged in.

Please refer to the description of the Octave command in this chapter for all procedures for entering octaves.

The Main Commands Menu

The Music Editor menus provide easy input and editing capabilities for the user of the MusicSystem. The Main Commands Menu is the first to appear after you enter the Music Editor. It is called the Main Commands Menu because it contains the commands that are most commonly used. These commands are:

1. Choice of Note duration
2. Dotted Note option
3. Measure Bars
4. Choice of Note Mode or Chord Mode
5. Rest option
6. Choice of accidentals
7. Change Pitch option
8. Change Duration option
9. Forward Delete, Backward Delete
10. Left cursor, Right cursor
11. Up to next part, Down to next part
12. Choice of other Menus
13. New File
14. Quit the Editor



Duration

Duration determines the relative length of a musical event, whether it is a note, a chord or a rest. The actual length of a note depends on the tempo of the composition. Once a duration is selected, it becomes the default duration. Duration applies to notes, chords, and rests. The duration is given as a numeric designation which is displayed on the Status Line. These values are detailed in the chart below.

Duration may be selected via light pen, paddle 1 or keyboard. The following chart shows the command keyboard command symbol for each designation.

Duration	Keyboard Command	Status Line
Whole Note	W	0
Dotted Whole	W.	1
Half Note	H	2
Dotted Half	H.	3
Quarter Note	Q	4
Dotted Quarter	Q.	5
Eighth Note	EI	6
Dotted Eighth	EI.	7
Sixteenth Note	S	8
Dotted Sixteenth	S.	9
Thirty-second Note	T	10
Dotted Thirty-second	T.	11

Dotted Option

The dotted option is used to extend the duration of a note, chord or rest by one half its specified value. A dotted note or chord appears in the score with a small dot to the right of the note, just as in standard musical notation. Note that the Dotted Option command modifies the selected duration, not the note itself. Thus, if the current duration is dotted, it will remain dotted until you change it by selecting a new duration value.

A dotted option is applied to a duration by choosing the "." symbol from the menu. From the keyboard, the command syntax is

. (RETURN)

If you wish to change duration to a "dotted duration" (of a **different** duration value), you must first specify the duration and then the dot. If you are simply adding a dot to the currently selected duration, you do not need to re-specify the duration; simply add the "dotted option". The default duration value will be a dotted note. This dotted duration value will remain until changed by selecting a new duration.

Measure

Measure is used to separate sections of a composition into logically defined segments. In traditional music, the measures in a composition are consistent in beat location from part to part. This is not necessary with the Music-System, but as always, good musical sense will be the overall guide. Measure bars have no effect on the sound or performance of the composition when it is played. See the GOTO command for details of forward or backward movement within each part of a composition.

A measure bar may be inserted at the current location of the cursor by choosing the Measure bar symbol from the Main Commands Menu. If entering Measure bars from the keyboard, the syntax is

M (RETURN)

Note Mode

Note Mode controls the automatic advancement of the cursor after each note is entered. If the Editor is set to Note Mode, the cursor advances after a note and pitch are selected.

Note Mode can be specified with the menus or the keyboard. From the menus, choose NOTE MODE. From the keyboard, the command syntax is

NOTES (RETURN)

The current mode is displayed on the Status Line.

Chord Mode

Chord Mode inhibits the automatic advancement of the cursor after each pitch in a chord is entered. If the Editor is set to Chord Mode, the cursor does not advance when a new pitch or accidental is selected. By moving the cursor with paddle 0 and selecting with button 0, more than one pitch can be inserted without the cursor advancing. Any selection **other than pitch, octave, RETURN, or accidentals** will cause the cursor to advance as in Note Mode.

Chord Mode can be specified by the menus or the keyboard. Via menu, choose CHORD MODE. If entering mode by keyboard, the command syntax is

CHORDS (RETURN)

The current mode is displayed on the Status Line.

All notes in a chord must be of the same duration, and must be entered successively. The entire chord must be entered before any additional musical events except accidentals, octave changes and RETURNs. For example, you may not enter a three-note chord, and later on add another note to it.

Rests

Rests mark places in the score where no music is to be played. The duration values for rests are the same as for notes.

The duration of a rest cannot be changed with the Change Duration command. If you attempt to do so, you receive the message

ITEM IS NOT A CHORD

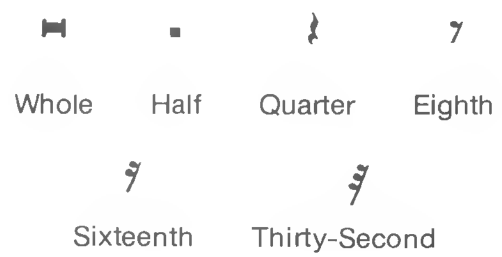
PRESS SPACE BAR WHEN READY

You must then press the space bar, delete the rest, and re-enter it.

Rests are inserted by choosing REST from the Main Commands Menu. The rest duration will equal the current duration. If a rest of different duration is desired, choose that duration before inserting the rest. To enter a rest from the keyboard, type

R (RETURN)

The symbols for each type of rest are as follows:



Rest is independent of paddle 0; there is no need to specify where on the staff lines the rest is to appear.

In the MusicSystem, Whole Rests always last for four beats no matter what the time signature. Whole Rests are given special treatment in the Music Editor. Whenever a Whole Rest is entered, a number appears in the score above the staff lines. This number represents the number of successive groups of four beats during which Whole Rests will be played.

If a single Whole Rest is entered, this number is "1". If more than one Whole Rest is entered without any intervening musical events, the number displayed is increased with each Whole Rest. For example, if three Whole Rests are entered in sequence, the number displayed is "3". This indicates that three times four beats (or 12 beats) of silence are played in this section of the score.

You may increase the length of an existing extended rest by moving the cursor to the right of the extended rest and (again) adding Whole Rests. The number displayed will increase with each Whole Rest entered.

Accidentals

Accidentals are added to notes (and notes in chords) to specify a change in the way that note is to be played. Accidentals consist of flats, sharps and naturals. Accidentals can only be entered as part of a note entry.

After entering a note that is to have an accidental, choose the desired accidental from the Main Commands Menu. It will appear to the left of the desired note. Place the cursor to the immediate right of the note or pitch to be modified and select the desired accidental. If you are entering accidentals from the keyboard, the command syntax is

accidental (RETURN)

Accidental Value	Keyboard Command
# (sharp)	#
## (double sharp)	##
b (flat)	*
bb (double flat)	**
(natural)	N

In the Music Editor, each note that is to be "accidentalized" must be **specifically and individually** indicated as such. This is different from conventional music notation in that, normally all notes of the same pitch in a measure following an accidentalized note will also be accidentalized. The Music Editor does not follow this convention.

Notation of Accidentalized Notes

The way the Music Editor displays accidentalized notes needs elaboration. Depending on the "sharped" or "flatted" status of the composition's key signature, the Music Editor will display all accidentalized notes as either flats or sharps, but will not display both within the same key signature. Follow these guidelines:

If a composition's key signature is natural, or has sharps, then all accidentals will be translated into sharps before they are displayed. For example:

In the key of C, if a note of E flat is entered, that note is translated into a D sharp, and displayed that way.

If a composition's key signature has flats, then all accidentalized notes are displayed as flats. Any sharps in the composition will be translated into flats before they are displayed. For example, in the key of F, D sharp is translated as an E flat and displayed that way.

Change Pitch

The CHNG PITCH option allows you to change the pitch of an existing note or chord. This option is simpler than deleting and inserting. The procedures for changing pitch are slightly different for notes and chords.

Notes

The note preceding the cursor is the one that will be affected. Place the cursor to the right of the note to be changed and select CHNG PITCH. The note is erased from the score. Now position the cursor to the staff line desired and press button 0 to insert the new pitch for that note. When used in NOTES mode, CHNG PITCH functions much like deleting a note and then re-inserting the new note.

Chords

In CHORDS mode, you need to specify which note within the chord is to be changed. Place the cursor to the right of the chord to be changed, and select CHNG PITCH. Position the cursor vertically and press button 0 to erase one note from the chord. Re-position the cursor to the desired staff line and press button 0 again. The new note will appear in the chord.

If you wish to only **remove** a note from the chord, do not press button 0 the second time.

If you are using CHNG PITCH from the keyboard, the syntax is

CP (RETURN)

to remove a note

CP n (RETURN)

to change the pitch of a note (n = new note's pitch)

CP n m (RETURN)

to change the pitch of a note in a chord (n = old pitch, m = new pitch)

Change Duration
Change Duration changes the duration of a note or chord. Since all the notes in a chord must be of the same duration, there is no difference in procedures for notes or chords. Change Duration changes the duration of an existing note or chord without re-entering the entire note or chord.

Select the desired "replacement" duration first. Place the cursor to the right of the note or chord, and select CHNG DUR. The duration of the note to the left of the cursor will be changed to the current default duration.

If entering CHNG DUR from the keyboard, the syntax is

CD (RETURN)

Forward Delete (→)
The Forward Delete arrow (→) deletes items in the score to the right of the cursor. When items are deleted, the screen is automatically adjusted.

Select Forward Delete. Holding the pen to the screen deletes notes continuously.

If you use Forward Delete from the keyboard, the syntax is

→ (no RETURN is needed for this command)

The REPEAT key can be used in conjunction with the keyboard Forward Delete to delete large sections of the composition. Use the REPEAT key cautiously as it moves very fast.

Backward Delete (←)
The Backward Delete arrow (←) deletes items in the score to the left of the cursor. When items are deleted, the screen is automatically adjusted.

Select Backward Delete. Holding the pen to the screen deletes notes continuously until you remove the pen from the screen.

If you use Backward Delete from the keyboard, the syntax is

← (no RETURN is needed for this command)

The REPEAT key can be used in conjunction with the keyboard Backward Delete to delete large sections of the composition. Use the REPEAT key cautiously as it moves very fast.

Move Cursor Left and Right
The left and right cursors move through each part of a composition. They are accessed by the Main Commands Menu or the keyboard. Cursor movement has no effect on the contents of the score. Left and right cursors cannot be used to advance out of the part currently being edited.

Select the → or ← boxes from the Main Commands Menu and hold the pen to the box for repeated cursor movement. When the pen is removed, the cursor movement stops.

If you use Left or Right Cursor Movement from the keyboard, the syntax is

CTRL-Q

for left movement (No RETURN is needed)

CTRL-A

for right movement (No RETURN is needed)

The REPEAT key can be used in conjunction with the keyboard Move Left or Move Right to move rapidly through the composition. Use the REPEAT key cautiously as it moves very fast.

Scroll Parts Up or Down
These cursors move out of the current part into the previous or following part. As the cursor advances up or down, the part displayed on the screen will change. Additionally, the Status Line will reflect the change in PART. When scrolling from part to part, the actual location in time will not change, but the measures displayed on the screen may not be consistent.

These cursors are accessed via the Main Commands Menu or from the keyboard. If used from the menu, hold the light pen to the appropriate box. The cursor will scroll up or down the parts until the pen is removed from the screen.

You can manipulate these cursors from the keyboard using Control (CTRL) characters. To move the cursor up one part, press

CTRL-W

To move the cursor down one part, press

CTRL-S

The REPEAT key can be used in conjunction with the CTRL commands to move rapidly through the composition. Use the REPEAT key with caution as it moves the cursor repeatedly and it's easy to accidentally go too far.

New
NEW clears the contents of the file currently in memory without QUITting the editor. It is useful if you have made many mistakes in a file and wish to start over. The Music Editor asks you to verify that you do indeed wish to clear the file away before it carries out the command.

Select NEW from the Main Commands Menu with the light pen or paddle 1. If you are using the keyboard, the syntax is

NEW (RETURN)

After entering the NEW command, either from the Menu or the keyboard, you are prompted with

VERIFY (Y/N) →

Type Y to complete the NEW command. The Music Editor buffer is cleared. Type any other key to cancel the NEW command.

Quit
QUIT is used to leave the Music Editor. The Music Editor will ask for verification before QUITting. Quitting the Music Editor causes the current contents of the composition file buffer to be lost forever. QUIT erases the high resolution composition screen and returns you to the Main Menu.

Select QUIT from the Main Commands Menu with the light pen or paddle 1. If you are using the keyboard, the syntax is

QUIT (RETURN)

You will then be asked to

VERIFY (Y/N) →

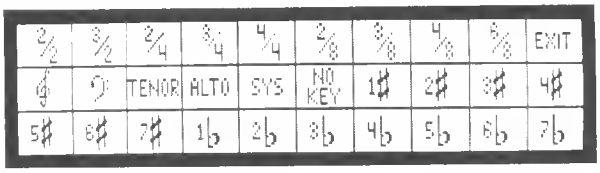
Type a Y to complete the command, or type an N if you have changed your mind.

Choosing Another Menu
You may only choose another menu from the menus. There is no way to choose another menu from the keyboard. Hold the light pen to the desired Menu box and that menu will appear on the screen. Optionally, you may use paddle 1 to choose another menu. To return to the Main Commands Menu, choose EXIT from the menu you are currently in.

The Signature Commands Menu

This menu contains all the commands needed to enter musical events dealing with the signatures, times and clefs that appear in a musical score. To get to this menu, choose SIG CMDS from the Main Commands Menu. The commands on this menu are:

1. Choice of Time
2. Choice of Clef
3. Choice of Key
4. EXIT to Main Commands Menu



A Note About Signature Displays
Signatures may be changed at any point in a composition. However, the portion of the score that can be displayed on the screen at one time does not always include the actual notation of the (possibly changing) signature. To resolve this, a section of the screen devoted to displaying the score is reserved for a continuous display of signatures.

The composing screen is designed so that the left-most portion of the screen is reserved for an ongoing display of the signature. This portion of the screen is separated from the "scrolling score" portion by a vertical bar. Remember that this portion of the screen is only a display, not a functional part of the sequential score.

The value of the signature displayed is always the same as the signature to the left of the current cursor position. Thus, the display will change each time the cursor encounters a different signature in the score.

When the cursor is at the beginning of the file, you will always see a treble clef, key of C, and

4/4 time signature display whether you put a signature in or not. These are the default signature values automatically provided by the Music Editor.

Clef
Clef designates the pitch values for each staff of a composition. You may specify one of five standard clefs, including the System Clef, which has both a treble and bass clef. The treble clef is the default clef. If you do not enter the clef as the first item in the score, the Music Editor will automatically provide a treble clef for you. Clef can be changed freely within each part in a composition.

Choose the SIG CMDS Menu. With the light pen or paddle 1, select one of the five clefs. If you attempt to enter something other than a clef as the first musical event in the composition, the Music Editor will automatically provide you with the treble clef.

From the keyboard, the command syntax for each clef is as follows:

Keyboard Command	Clef
BASS	bass
TREBLE	treble
TENOR	tenor
ALTO	alto
SYSTEM	system

You may change clef as often as you wish in a composition. Be aware that changing from System Clef to other clefs (and vice versa) will cause a somewhat different display of the staff lines in those portions of the score where the clef actually changes. No damage will be done to your composition; the only thing affected is the display. See the note about Key Signature Display in this chapter.

Key
Key determines which of the 15 major keys this part will be written in. The 15 standard keys may be selected from the SIG CMDS menu or from the keyboard.

If you do not enter a key value, the default of C is provided. However, no key is displayed on the score since that is the convention in regular music notation. If you enter the key of C with the menu symbol (NOKEY) or the keyboard command (KEY=0), a star appears in the score and the message "KEY = 0" is displayed on the Status Line. Simply using the default does not generate the star or the message.

Please see the note about Key Signature Display in this chapter.

Key may be changed at any time within each part of the composition. In the following chart, sharps are represented by "#", and flats are represented by "*".

Menu Symbol	Major Key	Relative Minor
No Sharps or Flats	C	(Am)
1 Sharp	G	(Em)
2 Sharps	D	(Bm)
3 Sharps	A	(F#m)
4 Sharps	E	(C#m)
5 Sharps	B	(G#m)
6 Sharps	F#	(D#m)
7 Sharps	C#	(A#m)
1 Flat	F	(Dm)
2 Flats	B*	(Gm)
3 Flats	E*	(Cm)
4 Flats	A*	(Fm)
5 Flats	D*	(B*m)
6 Flats	G*	(E*m)
7 Flats	C*	(A*m)

To enter one of the 15 keys, use the light pen or paddle 1 to select the desired key from the SIG CMDS menu.

If using the keyboard to enter a key, the command syntax is

KEY n * (RETURN)

where n is 1 - 7, or

KEY n # (RETURN)

where n is 1 - 7, or

KEY 0 (RETURN)

if there are no sharps or flats.

If you give a number that is out of range, you will get an error message. Press the space bar to recover. Then retype the correct number.

Time
The time signature is expressed by two numbers that appear as a fraction. The upper number represents the number of beats in each measure; the lower number represents the note duration assigned to one beat. For

example, in a 3/4 time signature, there are three beats per measure and a quarter note is assigned the duration of one beat. In the Music Editor, you can assign one of nine standard time signatures or you can create a time signature of your own.

If no time signature is entered before any notes or rests, the Music Editor will automatically supply a default time of 4/4. Please see the note on Signature Display in this chapter.

To enter a standard time signature, select one of the nine choices from the SIG CMDS menu. The time signature is normally entered before any notes, chords or rests and after the clef and key signature.

Enter non-standard time signatures from the keyboard. You may also enter standard time signatures from the keyboard. The syntax for keyboard entry of time signatures is

TIME nn/nn (RETURN)

The upper set of digits can be any number from 2 to 32. The lower set of digits must be 1, 2, 4, 8, 16 or 32. Entering a value that is not allowed for TIME causes the message

ILLEGAL TIMESIG

PRESS SPACE BAR WHEN READY

to appear. The TIME command is cancelled.

Additional time signatures may be specified at any point in the composition. Time signatures need not be consistent from part to part.

The Note Modifier Menu

This menu has all the modifiers that may be used to produce accents and emphasis on individual notes or groups of notes. Choose MOD CMDS from the Main Commands Menu. The commands on this menu are

1. Choice of note and chord accents
2. Choice of dynamic accents
3. No Accent — to remove an accent
4. Tie notes
5. EXIT to Main Commands Menu

.	▾	—	>	^					EXIT
sf	sff	sfff	fz	ffz	fffz	sfz	sffz	sfffz	NO ACNT
TIE									

Accents
Accents modify the way a note is played. There are various types of modifiers, each with a special purpose and function. More than one accent may be applied to a single note or single note in a chord; however, it is possible for the **display** of the accents to become distorted due to crowding on the screen. If more than one accent is applied to a note and the display is affected, all accents still are functional.

If an accent is applied to an item that is not a note or a chord, the message

ITEM IS NOT A CHORD

PRESS SPACE BAR WHEN READY

appears and the accent is not included in the composition.

Accents in the Music Editor include staccato, staccatissimo, tenuto, low percussive and high percussive marks. These accents may be applied to chords as well as notes. More than one accent may be applied to a note or chord by inserting them in sequence. Accents may be inserted following each note or may later be added to an existing set of notes.

To insert an accent, locate the cursor to the right of the note to be modified and use the light pen, paddle 1/button 1 or the keyboard. The vertical position of the cursor is insignificant when inserting accents. Use this chart to enter accents, either from the menu or from the keyboard.

Accent	**Keyboard Command**	**Menu Symbol**
Staccato	!	. (dot)
Staccatissimo	!!	v (wedge)
Tenuto	%	- (underbar)
Low percussive	↑ (up arrow)	→ (right arrow)
High percussive	↑↑ (2 up arrows)	↑ (up arrow)

If more than one accent is used for a single note or chord, the accent symbols will be printed on top of one another. This is normal.

Dynamic Accents
Dynamic accents may be applied to notes and chords in the same way as accents (above). A dynamic accent is a simultaneous combination of a temporary dynamic change and one of the elementary accents (above).

Dynamic accents may be inserted with the light pen, paddle 1/button 1 or the keyboard. Place the cursor to the immediate right of the note or chord to be modified and select the dynamic accent. If using the keyboard to enter dynamic accents, each accent command must be followed by a press of the RETURN key. The keyboard commands are as follows:

Accent	Keyboard Command
sf	SF
sff	SFF
sfff	SFFF
fz	FZ
ffz	FFZ
fffz	FFFZ
sfz	SFZ
sffz	SFFZ
sfffz	SFFFZ

Use of Dynamic Accents in Versions 2.0 and Earlier

The dynamic accents described above are fully integrated into the Music Editor as of Version 2.0. They have not yet been totally integrated into the functioning of the Music Player as of this version.

Currently, the "accent portion" of a dynamic accent is recognized by the Music Player, but the "dynamic" portion is not. Thus, the Music Player treats a dynamic accent like a regular accent.

You may include all dynamic accents in any compositions you create with the Music Editor, but they will not be "fully recognized" by the Music Player until a subsequent version of the Music Player software is released. When they are included in the Music Player, all compositions including dynamic accents will then feature the full effect of the dynamic accent during performance of the composition.

No Accent

No Accent removes an accent or dynamic accent from any note or chord without removing the entire note or chord. No Accent removes all accents from a note or chord; you may not use No Accent to remove one of a set of accents that are on a single note or chord.

Place the cursor to the immediate right of the accented note or chord. Use the light pen or paddle 1/button 1 to select No Accent. If you are using the keyboard to remove an accent, the syntax is

NOAC (RETURN)

Ties

TIE joins two successive notes of **the same pitch** to each other. The two notes may be of different duration. Using TIE, you can create a note or chord of any desired duration. For example, to create a note with the combined duration of a half note and an eighth note, use Tie. A tie is designated in the score by a "sideways bracket" above the notes being tied.

You cannot TIE two notes of different pitches. Do not attempt to simulate a slur in this way.

Place the cursor to the immediate right of the FIRST of the two notes to be tied. Select TIE from the MOD CMDS Menu via the light pen or paddle 1/button 1. If you are using the keyboard, the syntax is

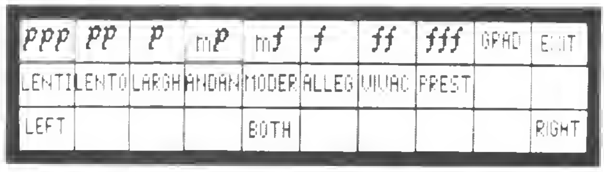
TIE (RETURN)

To erase a TIE, place the cursor between the two notes that are tied together and type TIE again. The Tie will be removed.

The Sound Control Menu

This menu provides the capability of applying dynamics to the score, and specifying tempo and spatial location in the first part of a composition. The commands on this menu are

1. Choice of Dynamics
2. Choice of Tempo
3. Choice of Spacial Location
4. EXIT to Main Commands Menu



Dynamics

Dynamics determine the relative loudness of the composition. In conventional music, a dynamic is indicated at the beginning of each part. Dynamics may be changed as often as desired within a part. They need not appear in consistent measures across the parts. A

dynamic remains in effect until a new dynamic is encountered. The Music Editor includes two types of dynamics: Absolute and Relative.

Absolute Dynamics: The dynamics available are, in sequence of softest to loudest

ppp, pp, p, mp, mf, f, ff, and fff

Relative Dynamics: A relative change in dynamics alters the loudness to a relative degree, based on the numeric value assigned to that dynamic. Relative dynamics are represented by a + or − sign and an integer from 0 to 127. Relative dynamics may only be entered via the keyboard.

Absolute Dynamics are inserted by placing the cursor to the immediate left of the first note to be affected by the dynamic. The vertical position of the cursor does not matter when inserting dynamics. Select the desired dynamic from the SOUND CMDS menu. If you are using the keyboard, the syntax for entering absolute dynamics is

Dynamic	**Keyboard Command**	**Absolute Value**
ppp	PPP	40
pp	PP	50
p	P	60
mp	MP	70
mf	MF	80
f	FT	90
ff	FF	100
fff	FFF	110

The designated absolute dynamic is displayed in the score and the value of that dynamic is displayed on the status line. The default absolute dynamic is MF.

The syntax for entering relative dynamics is

DYN + (or −) n (RETURN)

where n is an integer from 0 to 127. If you give a number that is out of range, you will get an error message. To recover, press the space bar and type the correct number.

When a relative dynamic is entered, a star appears in the score and the value of the relative dynamic is displayed on the Status Line.

Tempo

Tempo determines the fundamental speed at which all parts of the composition are performed. The tempo is consistent for all parts of the composition. Tempo may be changed any number of times within a composition, but it always changes consistently for all parts. To maintain this necessary integrity, the Music Editor only allows Tempo to be designated in Part 1 of a composition. If you attempt to enter a tempo mark in any part other than the first part, the message

MUST BE PART 1

PRESS SPACE BAR WHEN READY

is displayed. The Music Editor displays any tempo indications in all parts when they are made in Part 1.

Two types of tempos may be used in the Music Editor: Italian and numeric. Italian tempos may only be inserted from the menus, and numeric tempos may only be inserted from the keyboard.

Tempo items are designated in the score by stars and the value of a star is displayed on the Status Line when the cursor is located at that position in the score.

To enter a tempo indication in the score of Part 1, first be sure that you are editing Part 1. The current part is displayed on the Status Line. Place the cursor in the location where you want the tempo to be located. With the light pen or paddle 1, select the desired tempo indication. The choices of Italian tempos and their respective numeric values are, from slowest to fastest:

Italian Tempo	**Menu Symbol**	**Numeric Value**
lentissimo	LENTI	45
lento	LENTO	50
larghetto	LARGH	55
andante	ANDAN	60
moderato	MODER	90
allegro	ALLEG	120
vivace	VIVAC	165
presto	PREST	185

The default is moderato.

As with Italian tempo, you must be in Part 1 to enter a Numeric tempo. You may specify either an absolute numeric tempo or a relative tempo, which will increase or decrease the previous tempo setting.

To specify an absolute numeric tempo, the syntax is

TEMPO nnn (RETURN)

where nnn is a number from 040 to 200, slowest to fastest.

If the tempo is unspecified, the Music Player assigns a default value of 90.

To modify the tempo with a relative setting, the syntax is

TEMPO + (or −) nnn (RETURN)

where nnn is a number from 000 to 160, least to most amount of relative increase or decrease. If you give a number that is out of range, you will get an error message. Press the space bar to recover. Then type the correct number.

The Location of Tempo Marks

As indicated, tempo may only be specified in Part 1 of a composition. In addition to this necessary restriction, you must be sure that the location of a tempo mark does not coincide with a note in any of the parts. In other words, a tempo mark may only occur on **note boundaries**. For example,

Part 1 ♩#♩
Part 2 𝅗𝅥 "

is illegal, because the tempo mark in Part 2 occurs during the playing of the half note. To correct this, you can locate the tempo mark on a note boundary by changing the score of Part 2 as follows:

Part 2 ♩#♩ —by "tie"-ing 2 quarter notes to create a "quasi half note"

The Music Editor currently does not check for the presence of tempo marks occurring in places other than on note boundaries. However, the Music Player checks for the note-boundary alignment of tempo marks in all parts during the "compilation" of the composition file. If a tempo mark is found which does not occur on a note boundary, the Music Player displays the message

INVALID TEMPO ALIGNMENT

PRESS SPACE BAR WHEN READY

and the compilation is aborted. When you press the space bar the Main Menu is returned to the screen. You can then go to the Editor and review each part of the composition to determine which tempo mark is not on a note boundary.

In addition, you may get the INVALID TEMPO ALIGNMENT message when you try to use the Music Player to play a file that has been merged with the Music Merger. This will be discussed in detail in the chapter on the Music Merger.

Spatial Location

Spatial Location determines which speaker (or both) each part of a composition will come from. Spatial Location may be designated for each part, but it may not be changed within the part. It is entered at the beginning of the part, following the signature information for each part and before the first note, chord or rest. Spatial Location is designated by a star in the score.

You may include Spatial Location marks anywhere in a composition, but only the first mark at the beginning of each part will be recognized by the Music Player at this time. Later versions of the Music Player will recognize more Spatial Location marks.

Place the cursor to the right of the signature information at the beginning of each part, and select either LEFT, RIGHT or BOTH from the Sound Commands Menu. A star appears in the score, and the value of the Spacial Location appears on the Status Line. If no Spatial Location is provided, the default of BOTH is provided by the system. If you are using the keyboard, the syntax is

LOC x CR

where x is LEFT, RIGHT or BOTH, or where x is a number, 0 - 9.

When entering Spatial Location from the menu, the values displayed on the Status Line are

LEFT = 0
BOTH = 4
RIGHT = 9

If you enter a number that is out of range, you will get an error message.

If you enter a value for Spatial Location from the keyboard, the values become translated into Left, Right or Both designations according to the following chart:

Keyboard Entry	Music Player Definition
0	LEFT
1 thru 8	BOTH
9	RIGHT

Thus, you cannot specify a relative degree of Spatial Location because the Music Player will translate all values between 1 and 8 into an **equal Left and Right split**, and will translate 0 into all Left and 9 into all Right.

If no entry is made for the Spatial Location of a single part composition, the default is BOTH speakers.

If no entry is made for Spatial Location for a composition of more than one part, the parts are assigned RIGHT for PART 1, LEFT for PART 2, etc., until all parts are assigned a LEFT or RIGHT Spatial Location. Thus, all odd numbered parts are assigned RIGHT locations; all even numbered parts are assigned LEFT locations.

Non-Menu Commands

The following commands are not located on any of the four menus and may only be entered with the keyboard.

Load
The Load command places an existing composition file into memory for editing. Before loading any composition file, the Music Editor asks for verification, because loading removes any data that is currently in the Music Editor buffer.

To LOAD a file into memory, the syntax is

LOAD filename (RETURN)

The standard options for specifying drive and slot number may be used as in standard DOS instructions.

After you enter a filename, the Music Editor prompts you to

VERIFY (Y/N) →

Typing a Y continues the LOAD operation. Typing an N cancels the LOAD operation.

If the Music Editor cannot find the designated composition file, it displays the message

DISK ERROR: FILE DOES NOT EXIST

PRESS SPACE BAR WHEN READY

Check the directory of the disk containing the desired composition by typing CATALOG from the Main Menu. Re-enter the Music Editor and try LOADing again. This error message also occurs if you have designated a "PLAY." composition file, instead of a Comp file. The Music Editor will not allow you to load Play files.

The only time you do not need to LOAD a file into the Music Editor is when you are beginning the creation of a new composition. Simply begin entering the required musical events, i.e., the Key Signature information.

Save
SAVE is used to save any changes or entries that have been made to a file since the previous SAVE command. It is a good editing practice to SAVE the file you are working with often to protect your work from accidental deletion. SAVE does not affect the contents of the file buffer, it only writes the contents to disk.

To SAVE a file, the syntax is

SAVE filename (RETURN)

If you attempt to SAVE to an existing Comp file, the message

FILE EXISTS AND
WILL BE OVERWRITTEN

VERIFY (Y/N) →

appears. Press a Y to overwrite the old file. Press an N to cancel the SAVE.

If you attempt to SAVE to one of the System diskettes, the message

DISK ERROR: DISK FULL

PRESS SPACE BAR WHEN READY

appears. You cannot SAVE Comp files to either of the System diskettes.

Pitch
You can designate a specific pitch for a note from the keyboard if you do not wish to use the paddle 0/button 0 technique for entering notes. All notes entered this way will be placed in the octave that is currently displayed on the Status Line. To enter a note outside of that octave, you must first change the octave with the Octave command.

To enter a note from the keyboard, first choose the desired mode, Notes or Chords to control the automatic advancement of the cursor. Determine that the octave displayed on the keyboard is the octave you want the note to be in. Then type the note desired. The allowable note values are

C, D, E, F, G, A, B

Octave
Octave allows you to specify the octave into which all notes and chords **entered from the keyboard** will be placed. By specifying an

octave value from the keyboard, you can control the placement of notes and chords when your paddles are not plugged in. After specifying an octave value with the OCTAVE command, you may still enter notes or chords **outside the range of that octave** by changing the current pitch/octave value with paddle 0, and inserting the note or chord with button 0.

The octave value designated in the OCTAVE command is displayed on the Status Line, and is not affected by changing the octave with paddle 0. Please see the Tutorial dealing with Octaves and the section of this chapter called "A Note About Octaves" for full details of the way the MusicSystem controls the use and display of octaves.

To assign a particular octave for entering notes from the keyboard, use the OCTAVE command. Use of OCTAVE does not restrict the use of paddle 0/button 0 to enter notes and chords outside the octave specified. The syntax for octave is

:n (RETURN)

where n is the value of the desired octave, 0 - 7.

The default value for octave is 4.

The capability for changing the octave value with this command allows you to enter notes and chords into different octaves even if the paddles are not plugged in.

GOTO

GOTO is used to "go to" any specified measure or to the beginning or end of the part currently being edited. If no measure bars are used in the composition, you may only go to the beginning or end of the part. GOTO may only be used from the keyboard.

To implement GOTO, type

GOTO n (RETURN)

where n is the number of the measure you wish to move to. The values allowed are 0 to 255.

Several special situations exist in use of the GOTO command:

GOTO beginning or end of part:
To go to the beginning, type

GOTO 0

To go to the end, type

GOTO 255

GOTO in compositions without measure bars:
If you have not inserted measure bars, you may only GOTO the beginning or end of the part. GOTO 0 or GOTO 1 moves the cursor to the beginning of the part (measure 1 is implied, even if no measure bars are inserted). GOTO 2 (or any number larger than 1) moves the cursor to the end of the part.

If you specify a measure number greater than the last sequentially numbered measure in the part, the cursor will be located at the end of the part.

Entering a number greater than 255 in a GOTO command produces the message

INTEGER 255 NOT ALLOWED

Sync

An extremely powerful feature of the MusicSystem is its ability to synchronize external events with music being played. Sync commands to a secondary program may be used to "trigger" slide projectors, video tape players, or other secondary programs while music is playing. Although this feature is not fully implemented at this time, MusicSystem owners should be aware of this powerful feature.

Synchronization marks specify a location in the score that can be accessed by user-written programs. They are designated by stars in the score. When the cursor is on a sync mark, the value of that mark is displayed on the Status Line.

To insert a sync mark, type

SYNC nnn (RETURN)

where nnn is a number from 0 to 255.

Use of SYNC in Versions 2.0 and Earlier

The Sync command described above is not fully implemented in the MusicSystem as of Version 2.0. It is functional in the Music Editor and Sync "setups" can be entered, but no action can be taken in Music Player as of this version.

Instrument Assignment

Instruments are assigned for each part of a composition by using this command. Instruments must be assigned at the beginning of the score for each part, before notes or rests, and only one instrument may be assigned for each part. Instrument assignments may not be changed within a part. The instrument named must be among the set of instruments provided with the MusicSystem. The instrument assignment is designated in the score by a star. When

the cursor is on the star, the name of the instrument is displayed on the Status Line.

To assign an instrument to a part, place the cursor at the beginning of the score following the signature information and before the first note, chord or rest. Then type

INST instrument name (RETURN)

If no instrument is assigned, the default instrument is ORGAN.

Add Part

Adding a new part is begun by using this command. Add Part cannot be used to create the first part. You **must** specify a part name when using Add Part. If left blank, a syntax error message occurs.

To add a new part, type

ADDP part name (RETURN)

All part names must begin with an alphabetical character, and may not be longer than eight characters or include commas. Entering more than eight characters in a part name is allowed, but the ninth and subsequent characters will be truncated. No more than 16 parts may be used in any composition.

You may use duplicate part names for different parts, but this is not recommended. If you do choose duplicate names for different parts, you will need to reference their respective part numbers if and when you choose to delete any of those parts. See the DELETE PART command for details.

Delete Part

Deleting a part removes the names part from the composition. You must supply a part name or part number when using the DELETE PART command. Entering DELP without a part name or with a name or number that is not in the composition produces the message

PART NOT FOUND

PRESS SPACE BAR WHEN READY

and the deletion is cancelled.

To delete a part, type

DELP part name (RETURN)

or

DELP part number (RETURN)

After entering the DELP command, you are prompted with

VERIFY (Y/N) →

Type a Y to finish the deletion, or type an N to cancel the Delete Part command.

Print

The Print command allows you to immortalize on paper the musical scores you have created with the Music Editor. You can print all or part of a composition in one of two sizes.

If you want to print a Comp file, and if you have a Silentype plugged into slot #1, from the Music Editor Main Commands Menu type

PRINT (RETURN)

This will cause the file currently in the Music Editor to be printed in its entirety. All parts and all measures in each part will be printed in the octave indicated on the Status Line in single size. If you want to print only part of a composition, or if you want to change the octave or the size in which it is printed, you can use the Print Options discussed below.

The Print Options

There are five print options available with the Print command. You can use any combination of these options in any order. If you don't specify one of the options, the default for that option will be assumed. The options are

- PT for part number or range
- M for measure number or range
- S for printer's slot number
- O for octave number
- DB for double size printing

The Part and Measure options can be given either an integer or a "range". Range is designated by two numbers separated with a slash (/). For examle, to print a range of parts, say parts 2 through 4, you would type

PRINT PT 2/4

The 2/4 is called the "complement" of the Part option. The Part option lets you print all parts of a composition, or specify which parts you would like to print. Part takes either an integer or a range complement. The default for the Parts option is all parts.

The Measure option lets you specify which measures you would like to print. Measure takes either an integer or a range complement. The default for the Measure option is all measures.

The Slot Number option lets you specify which slot your printer is plugged into. Slot Number

takes an integer complement. The default for the Slot Number option is 1. If your printer is not plugged into slot number 1, you must specify the slot number each time you print.

NOTE: If you give the wrong slot number, the program will "hang", and you will probably lose anything stored in memory, including the song you are about to print! To recover, you must re-boot the System 1 diskette.

The Octave option lets you print a composition in different octaves. This is especially handy if the octave your composition is written in does not show up on the screen. Octave takes an integer complement. The default for the Octave option is the octave which appears in the Status Line, or, if your paddles are plugged in, the octave which appears as the Paddle Octave.

The Double Size option lets you print your composition in a special large size for easy reading. Double Size does not take a complement. The default for the Double Size option is Single Size.

The syntax for the Print command and its options is as follows:

PRINT PT n/n, M n/n, S n, O n,
DB (RETURN)

There must be a space between "PRINT" and the first option. Commas must be used between options. The spaces between options are optional.

Each option can be specified only once in a Print command. If you try to specify an option twice, you will get a Syntax Error. If you try to print a part of a composition that doesn't exist, for example, if you specify M 16 when there are only 12 measures in the composition, the message

NO SCORE PRINTED

PRESS SPACE BAR WHEN READY

will appear on the screen, and the printer will not print. When this occurs, you can retype the Print command, this time with the correct options. If you don't wish to try printing again, you can type any Editor command.

Chapter VI
The Music Merger

With the Music Merger you can produce large Comp files by combining two smaller Comp files you entered with the Music Editor. In this way you can play compositions that are too long for the Music Editor to handle. The Music Merger also lets you copy Comp files. This is the only way you can transfer Comp files, since they can't be BLOADed or BSAVEd.

To access the Music Merger, go to the Main Menu on the System 1 diskette. Then choose option 3, Music Merger. The screen will display

** WAIT — LOADING **

and then, when the Music Merger is loaded,

MOUNTAIN COMPUTER —
THE MUSIC MERGER

ENTER 1ST COMP FILENAME
→

appears on the screen.

Merging Compositions

The merging of compositions is done in three easy steps. This section will tell you how.

Step 1
When the prompt for the first filename is displayed on the screen, you have two options. You can either

- leave the Music Merger and return the Main Menu to the screen by pressing CTRL-C,
- or you can type the name of the first file to be merged.

To begin the merge process, type the name of the first of the two Comp files you wish to combine. If the diskette that contains the file is not in the boot drive, you must specify the slot and drive numbers. For example, if the Comp file is in drive 2, slot 5, and called COMP.BLUES you would type

BLUES, S5,D2

and press the RETURN key. If the file you specified does not exist on that diskette, the message

DISK ERROR: FILE DOES NOT EXIST
FILENAME: COMP.BLUES
INSERT COMP DISKETTE INTO S5, D2
SPACE BAR WHEN READY
OR ESC TO CANCEL

appears on the screen. You then have two options:

- Insert the appropriate diskette and press the space bar. If the Music Merger is still unable to find the Comp file, the screen will display the same message again.
- Press the ESC key to return the first Comp filename prompt to the screen.

Step 2
When the first filename has been accepted, the Music Merger prompts you for the name of the second Comp file to be merged:

ENTER 2ND COMP FILENAME
(CR TO COPY)
→

- To disregard the first Comp filename, press the ESC key.
- To continue with the merge process, type the name of the file you wish to append to the first one.

If you have a one-drive system, both Comp files to be merged must be on the same diskette. If you have a two-drive system, both Comp files must be in the system at the same time. That is, each file must be on a diskette that is in one of your disk drives.

If the Music Merger can't find the second Comp filename, for example, if you mistyped it, you will get the message

DISK ERROR: FILE DOES NOT EXIST
FILENAME: COMP.BLUES

PRESS SPACE BAR WHEN READY.

When you press the space bar you will be prompted to type the second Comp filename again.

Step 3
After it finds the second Comp file, the Music Merger asks for the name under which you wish to store the new merge file:

ENTER MERGE COMP FILENAME
→

- If, at this point, you press the ESC key, the names you typed for the first and second files will be disregarded and the Music Merger will start over again, prompting you to type a new first filename to be merged.
- To complete the merge process, type the name you wish to use. If a Comp file of that name already exists, you will get the message

FILE EXISTS AND
WILL BE OVERWRITTEN.
VERIFY Y/N →

NOTE: The filename you give for the newly merged file cannot be the same as either of the Comp files being merged. If you try to give it one of those names, you will get an error message.

When the new filename has been accepted by the Music Merger, the merging of the two files begins. You will see this message displayed on the screen:

** MERGE IN PROGRESS **

After a few minutes (large Comp files take longer) the merge will be complete. The screen will then display

** MERGE COMPLETE **

PRESS SPACE BAR WHEN READY

When you press the space bar, the Music Merger will be ready to start over again, prompting you to type a new first filename. If you don't want to merge any more files, press CTRL-C to return the Main Menu to the screen. You can then type

CATALOG

to verify that your files were properly merged. If the process was completed, the name you gave for the new merged file should appear on the diskette catalog. This is the file you will use in the Music Player to create a Play file.

NOTE: The Music Merger uses regular Apple DOS syntax for specifying slot and drive numbers. Just like the Apple DOS, you can generate "FILE DOES NOT EXIST" or "FILE ALREADY EXISTS" messages by specifying wrong slot and drive numbers.

The Music Merger can merge only two files at once. You can combine several small Comp files to create one large file by repeatedly merging new files onto the same Comp file. The maximum allowable size for a composition is difficult to determine because "size" is based on a complex formula of length, complexity, number of parts, instruments, and other factors. Whenever you are merging multiple files, consider compiling the merged files in the Music Player after every few merges to be sure that the Music Player can handle a file of the size you are creating.

Some Useful Details

The Music Merger combines Comp files on a part to part basis. Thus Part 1 in the first file will be attached to Part 1 in the second file. The Part names in the first file will be used in the newly merged file; the Part names in the second file will be disregarded.

The maximum polyphony size for each part is affected by the merge process. ("Polyphony" is the number of notes played at one time, as in a chord. This concept is discussed in much more detail in the chapter on Instrument Definition.) If one of the parts in one of the Comp files to be merged has a larger maximum polyphony size than the corresponding part in the other Comp file, the larger of the two sizes becomes the maximum polyphony size for both parts. Polyphony is discussed in full detail in the chapter on Instrument Definition.

You cannot edit a merged file if it has reached a size that will not fit in the Music Editor. For this reason it is advisable to keep unmerged versions of all your compositions.

In some instances tempo marks in a composition to be merged can cause problems. If the first file has fewer parts than the second, and the second file has a tempo mark, the two files will seem to merge normally, but, when you try to play the newly merged composition with the Music Player you will get the message

INVALID TEMPO ALIGNMENT

To resolve this problem, add extra parts to the shorter first file, filling the extra parts with rests.

Copying Compositions

Not only can you merge Comp files with the Music Merger, but you can copy them, too, in three easy steps. You can copy Comp files from one diskette to another, or you can copy the same file to a new name on the same diskette. The copying feature is handy for backing up your work, or for when you want to keep many versions of a composition on a diskette, each version varying only slightly.

Step 1

You begin the copying process the same way you begin a merge. First choose the Music Merger from the Main Menu. Then you will be prompted to

ENTER 1ST COMP FILE NAME
→

just as if you were getting ready to merge two files.

Type the name of the file you wish to copy, using the same syntax as was used for the first merge file. If the file does not exist, or if you mistyped it, the message

DISK ERROR: FILE DOES NOT EXIST
FILENAME: COMP.XXXXX
INSERT COMP DISKETTE INTO Sn, Dn
SPACE BAR WHEN READY
OR ESC TO CANCEL

(where XXXXX is the filename you specified, and n = the slot and drive numbers) will appear on the screen. You could then either

- insert the correct diskette and press the space bar, or
- press the ESC key to return to the first filename prompt.

Step 2
When the first filename has been accepted, the Music Merger will prompt you to

ENTER 2ND COMP FILENAME
(CR TO COPY)
→

At this point you can

- press the ESC key to eliminate the first filename, in which case you would be prompted for a new first filename,
- or press the RETURN key to indicate that you wish to copy the file.

Step 3
After you have pressed RETURN, the screen will display

ENTER COPY COMP FILENAME
→

- If you wish to cancel the copy operation, you can press the ESC key to go back to the prompt for the first filename.
- If you wish to continue the copy process, type the name under which you wish to save the new copy of the Comp file. If you want to save the file onto a different diskette, you should include slot and drive numbers, or switch diskettes. The name of the new copy can be the same as that of the old copy.

If you type a filename, thus indicating that you wish to continue the copy process, you will be prompted to

INSERT COPY DISKETTE INTO Sn, Dn

PRESS SPACE BAR WHEN READY.

When the diskettes are in their proper places, the message

** COPY IN PROGRESS **

will appear on the screen. In a moment the copy will be complete, and the screen will display

COPY COMPLETE

PRESS SPACE BAR WHEN READY.

When you press the space bar you will be prompted for a new first Comp filename. You may want to check that the Comp file actually was copied. Press CTRL-C to display the Main Menu, and then CATALOG the diskette on which your new copy should be. The filename you gave in step 3 should be displayed in the diskette catalog.

Chapter VII
The Music Player

How It Works — An Overview

The Music Player performs the compositions created in the Music Editor. It allows special changes to the way the composition is performed each time it is played. This chapter provides detailed instructions for using the Music Player. No separate tutorial is required because the operation is simple and follows a set of "prompts" which guide you through the steps to use it. The sample screens used in this chapter reflect, in part, the composition created in the Music Editor tutorial.

Capabilities

The Music Player allows you to:

- convert the COMP files created in the Music Editor into Play files used by the Music Player to perform a song.
- change the instruments assigned to each part of the composition when the composition is played.
- re-assign the speaker location for each part in the composition or choose an option for monohphonic performance.
- cancel a performance at any point while playing.
- save a song either before or after hearing it played.

The capability to change the instrument assignment and spatial location from a special Player Menu is a valuable tool when developing a complex multi-part composition. Rather than determine these characteristics from within the Music Editor, you can freely experiment with them in the Music Player and when a pleasing combination is found, make "permanent assignments" for the composition by saving the Play file.

The Player Menu allows you to play a song, change instruments or speaker locations, set all parts in the song to mono, and save the song. It looks like this:

```
CHOICES: RETURN = PLAY,
  1 = CHANGE INST, 2 = CHANGE SPKR
  3 = SET ALL TO MONO, 4 = SAVE FILE;
  TYPE YOUR CHOICE
```

Comp Files into Play Files

All song files that have been created by the Music Editor are prefixed with "COMP.". Before these Comp files can be played by the MusicSystem, they must be processed or converted into Play files; the output of the conversion is a "PLAY." file.

This conversion of the Comp file is a two-step process; first the song is compiled, and then the instrument definition files (and their waveform files) are "bonded" to create the resulting Play file. These Play files are then used by the Music Player to play the song.

This second step of the conversion (the bonding step) only occurs the first time you compile a song or when you change one or more of the instruments used by the song when you play or save it.

When you change instruments in a song, you will need to re-bond the instrument definition files (IDEF.) and the waveform files (WAVE.) associated with the newly assigned **set** of instruments to the song being processed.

Combinations of Comp and Play Files

The procedures and steps for using the Music Player depend on the type of file you supply to the Music Player and on the instrument and waveform files associated with each song. Depending on whether you have ever saved a Play file with the Music Player, a composition could have the following combinations of files associated with it:

Comp File Only

If a composition has been created in the Music Editor, but has **not been processed by the Player**, you will only have a Comp file associated with that composition.

During the two-step conversion process, you will first compile the song and then "bond" the instrument definition files to the Play file.

Comp File and Play File

If a composition has been created in the Music Editor and then **processed and saved** in the Music Player, you will have both a Comp file and a Play file associated with that song. You will have an opportunity to choose which of these files will be used by the Music Player.

If you choose the Play file and you change instruments in the song, you must re-bond the new instruments used in the song. This creates a new version of the Play file.

Play File Only
If a composition has been saved **under a name different from its Comp file name**, you will only have a Play file associated with that composition.

As in the above cases, any change to instruments used in the song will require the second "bonding" step to be performed.

Depending on the existence of the Comp and Play files, and the location on diskette of the instrument definition files associated with each composition, the system will display different screens, differing prompts and offer different processing options. These procedures are described in the following paragraphs, along with samples of the screens that are displayed by the Music Player.

Entering the Music Player

With one of the Main Menus (System 1 or System 2) on the screen, select the Music Player by

1. holding the light pen to the "1", or
2. moving the small cursor to the "1" and pressing button 0, or
3. typing "1" on the keyboard.

After selecting the Music Player from the Main Menu, you see:

** WAIT — LOADING **

This indicates that the Music Player programs are being loaded into memory. After a brief pause, this disappears and you see:

MOUNTAIN COMPUTER —
THE MUSIC PLAYER

ENTER COMP OR PLAY FILE NAME
(CTRL-C TO EXIT TO MAIN MENU)

NAME:

Enter the name of a COMP. or PLAY. file and press RETURN. If you are using a single drive system, you are then prompted:

INSERT PLAY OR COMP DISKETTE
INTO S#,D#
PRESS SPACE BAR WHEN READY

If you are using a **dual drive system**, and you place the PLAY or COMP diskette in a secondary drive (i.e., slot 6, drive 2), you must include the slot and drive designation when asked for the name of the song. If you do, the above prompt does not appear.

Observe all the standard DOS conventions for designating the disk drive with your composition diskette. The Music Player prompts will indicate where to place the diskette based on your inclusion (or exclusion) of slot and drive information in the response to the song name prompt.

At this point, if you enter thename of a composition **not found on the diskette**, the following message appears:

FILE : filename, S#, D#
** ERROR ** NO PLAY OR COMP FILE
PRESS SPACE BAR WHEN READY

You are then re-prompted for the filename.

Check your spelling of the composition name as displayed on the screen or do a CATALOG command from the Main Menu to see if the file is on the diskette. Try again.

Processing Songs with Comp Files

If a Comp file has never been processed by the Music Player, you will only have a Comp file associated with it. The Music Player will load the file and immediately begin the first step of the conversion process, **compiling it**. You will see a screen that looks something like this:

SONG: songname

PART #	**NAME**	**INST NAME**	**SPKR**
1 (1)	PART 1		
2 (1)	PART 2		

*** WAIT — COMPILING **

During the compile, the diskette will periodically be accessed. When the compile step is done, the Player Menu and the instrument and speaker designations are added:

SONG: songname

PART #	**NAME**	**INST NAME**	**SPKR**
1 (1)	PART 1	BRASS	RIGHT
2 (1)	PART 2	ORGAN	LEFT

CHOICES: RETURN = PLAY,
1 = CHANGE INST, 2 = CHANGE SPKR,
3 = SET ALL TO MONO, 4 = SAVE FILE;
TYPE YOUR CHOICE

Bonding the Instruments to the Song
At this point, you may play the song, change instruments, change speakers, set all parts to monophonic or save the file. We have not yet created a Play file; we still need to bond the instrument definition files to the compiled

Comp file. **Bonding is begun by choosing to play a song (hitting RETURN) or by choosing to save the song to disk (choosing 4 from the Player Menu).**

All the instrument definition files associated with each instrument for this song must be available for the bonding process to complete the creation of a Play file. When you choose to play the song or save it, you are prompted:

INSERT IDEF DISKETTE INTO S#, D#
SPACE BAR WHEN READY
OR ESC TO CANCEL

If these instrument definition files are on the same diskette as the song being processed, the bonding step continues when you press the space bar. When the bonding is complete, the song will play or be saved (following your response to a prompt), depending on your choice from the Player Menu.

If the instrument definition files are not found, you are again prompted to insert a diskette with the instrument definition files in a designated drive. This prompt will reappear until a diskette with the instrument definition files is inserted or the process is cancelled via ESC.

During the bonding step, the Music Player searches for each of the instrument definition files. If it finds the IDEF. file, it looks for the associated waveform (WAVE.) files that contribute to making the instrument definition. It works like this:

During the bonding step, the Music Player begins the process of determining the oscillator assignments by:

1. Looking on the diskette for IDEF. filename1 for instrument 1. If it finds it, it begins looking for the WAVE. files for instrument 1. If it finds them, the bonding continues. If it doesn't find either the IDEF. or WAVE. files, it prompts:

   DISK ERROR: FILE DOES NOT EXIST
   FILE NAME: yyyy.zzzzzz
   INSERT yyyy DISKETTE INTO S#,D#
   PRESS SPACE BAR WHEN READY
   OR ESC TO CANCEL

In this message, yyyy is either IDEF or WAVE; zzzzzz is the instrument name or waveform name. The filename displayed on the screen is the instrument file that was not found; it appears when **either the IDEF. or WAVE. files** for an instrument are not found.

You may insert other diskettes to attempt to find the files, or if you wish to cancel the bonding, press ESC. This causes a return to the Player Menu.

2. If all the IDEF. and WAVE. files for the first instrument are found, the Music Player repeats the above process for all other instruments used in the song. As before, if an IDEF. OR WAVE. file is not found, the above Disk Error message is issued. When the bonding is done, the song either will play or will be saved (following your response to a prompt).

After the Bond is Over

If you play the song, and it has not been saved before, after it plays, you are prompted:

SONG WILL BE SAVED
VERIFY (Y/N) →

If you want to keep this compiled composition, bonded with the existing instrument definition files, type Y. If not, type N. Typing Y causes this prompt to appear:

--- SAVE PLAY FILE ---
NAME :

If you play the song on a single drive system, after the song plays, you are prompted:

INSERT SYSTEM DISKETTE INTO S#,D#
PRESS SPACE BAR WHEN READY

Saving the Composition

You may save the newly processed song before or after playing it, but if you **do not save the file**, you will need to **re-convert** it each time you want to play it. If you do save the file, you can experiment with the speaker and instrument assignments without going through the compile step of the conversion process.

To begin the SAVE, choose "4" from the Player Menu.

If you save the song **before playing it**, you will see the prompt:

--- SAVE PLAY FILE ---
NAME :

Enter the name under which you wish to save the processed song. It can be identical to the Comp or Play name or it can be different.

Whether you save to a new name, or use the existing name, you are prompted

INSERT IDEF DISKETTE INTO S#,D#
SPACE BAR WHEN READY
OR ESC TO CANCEL

After the bonding step takes place, you are prompted

INSERT PLAY DISKETTE INTO S#,D#
PRESS SPACE BAR WHEN READY

The song is saved.

A Note on Filenames When Saving a Song
If you save the file under a filename that already exists, you will see:

FILE EXISTS AND
WILL BE OVERWRITTEN

VERIFY (Y/N) →

If you hit N, nothing is saved, and nothing is lost; you are returned to the Player Menu. If you hit Y, the new file's contents are saved, overwriting the old file. You may then proceed to play the song.

If you save it under the original name, you will then have both a Comp and Play file associated with that composition.

If you play a song that has **never** been saved, when the song is finished playing, the Music Player will prompt you

SONG WILL BE SAVED.

VERIFY (Y/N) →

Pressing Y issues another prompt:

--- SAVE PLAY FILE ---
NAME :

Enter the name that you wish to save this song under.

Changing Instruments

The Music Player has a series of prompts to guide you through any changes to instrument assignments. Simply follow the instructions that appear on the screen after choosing "1" from the Music Player Menu's options:

CHOICES: RETURN = PLAY,
1 = CHANGE INST, 2 = CHANGE SPKR,
3 = SET ALL TO MONO, 4 = SAVE FILE;
TYPE YOUR CHOICE

After typing "1" to begin an instrument change, the prompt

--- CHANGE INSTRUMENT ---
PART # TO CHANGE:

appears on the screen. Enter the number of the part in which you want to make an instrument change and press RETURN. Typing a number that is not a part number or a non-numeric character produces the message:

** BAD PART NUMBER **
PART # TO CHANGE:

Re-enter the part number. You are then prompted:

NEW INST:

Enter the name of an instrument from the list of instruments provided with the MusicSystem or one of your own creation.

To escape from this choice, press ESC at either the part # or instrument prompt. The instrument change is cancelled. You may re-attempt the change if you wish. When a change is complete, the new instrument name is displayed on the screen in the column labeled "INST NAME".

Entering an instrument that does not exist (i.e., no IDEF. and WAVE. files) is possible, but when you attempt to play or save the song, you will get the error message:

DISK ERROR: FILE DOES NOT EXIST
FILE NAME: yyyy.zzzzzz
INSERT yyyy DISKETTE INTO S#,D#
PRESS SPACE BAR WHEN READY
OR ESC TO CANCEL

The Music Player cannot find the needed instrument definition files. You may insert a diskette with the files in the designated drive or you may escape from this choice by pressing the ESC key.

See the section of this chapter called "Processing Songs with COMP. Files" for information on providing IDEF. and WAVE. files when you change instruments.

Changing Speaker Location

The Music Player has a series of prompts to guide you through any changes to speaker locations. Simply follow the instructions that appear on the screen after choosing from the available options:

CHOICES: RETURN = PLAY,
1 = CHANGE INST, 2 = CHANGE SPKR,
3 = SET ALL TO MONO, 4 = SAVE FILE;
TYPE YOUR CHOICE

After typing "2" to begin a speaker change, the prompt

--- CHANGE SPEAKER ---
PART # TO CHANGE:

appears on the screen. Enter the number of the part in which you want to make a speaker change and press RETURN. Typing a number that is not a part number or typing a letter produces the message:

** BAD PART NUMBER **
PART # TO CHANGE:

In this case, re-enter the part number. You are then prompted:

NEW SPEAKER (L, R, OR B):

Enter either L, R or B. Entering anything else causes the bell to sound on the Apple speaker. You are again prompted to enter either L, R or B.

To escape from this choice, press ESC at either the part # or speaker prompt. The speaker change is cancelled. You may re-attempt the change if you wish. When a change is complete, the new speaker location is displayed on the screen in the column labeled "SPKR".

Setting All Parts to Mono

You can set all parts of a song to MONO by choosing "3" from the Player Menu:

CHOICES: RETURN = PLAY,
1 = CHANGE INST, 2 = CHANGE SPKR,
3 = SET ALL TO MONO, 4 = SAVE FILE;
TYPE YOUR CHOICE

After pressing 3, you are prompted

— CHANGE ALL SPKR TO MONO —

VERIFY (Y/N) →

Pressing Y changes all parts to mono, N cancels the command.

Comp and Play Files of the Same Name

If you enter a composition name that has previously been processed and saved by the Music Player, the system will need to determine whether you want to play the compiled and bonded Play file or to process the source Comp file. After entering the composition name, you see:

THE PLAY FILE WILL BE LOADED
INSTEAD OF THE COMP FILE

VERIFY (Y/N) →

Pressing Y loads the Play file, pressing N loads the Comp file. Any other response causes the bell to sound on the Apple speaker; you will again be asked to verify that the Play file should be loaded.

If you press N, the Comp file is loaded and the compile step of the conversion begins:

SONG: song name

PART #	NAME	INST NAME	SPKR
1 (1)	PART 1		
2 (1)	PART 2		

*** WAIT — COMPILING **

Since this song is in the process of being changed from a Comp file to a Play file, you will need to supply the necessary IDEF. and WAVE. files for the Player to bond to each instrument.

If you press Y, the PLAY. file is loaded, the Music Player then displays

SONG: song name

PART #	NAME	INST NAME	SPKR
1 (1)	PART 1	BRASS	RIGHT
2 (1)	PART 2	ORGAN	LEFT

CHOICES: RETURN = PLAY,
1 = CHANGE INST, 2 = CHANGE SPKR,
3 = SET ALL TO MONO, 4 = SAVE FILE;
TYPE YOUR CHOICE

As before, either choosing to play the song or save the song begins the process of bonding the instrument definition files to the song. See the section of this chapter called "Processing Songs with COMP Files" for details of the bonding step.

Playing a Composition Which Only Has a Play File

If you enter the name of a composition that only has a Play file associated with it (i.e., was created by saving under a new name a previously processed Comp file), you will see:

YOU HAVE ONLY A PLAY FILE
BY THAT NAME

The screen clears and then displays the following:

SONG: song name

PART #	NAME	INST NAME	SPKR
1 (1)	PART 1	BRASS	RIGHT
2 (1)	PART 2	ORGAN	LEFT

CHOICES: RETURN = PLAY,
1 = CHANGE INST, 2 = CHANGE SPKR,
3 = SET ALL TO MONO, 4 = SAVE FILE;
TYPE YOUR CHOICE

You may play the song, change the speaker or instrument assignments or re-save the composition as described previously.

See the section of this chapter called "Processing Songs with Comp Files" for details of changing instruments in Play files.

Cancelling a Performance

To stop the Music Player at any time, type CTRL-C.

Depending on what you are doing when you type CTRL-C, the following will occur:

Function Interrupted	You Arrive At
1st Input for Song Name	Main Menu 1 or Main Menu 2
Conversion	1st Input for Song Name
Player Menu	1st Input for Song Name
Playing Songs	1st Input for Song Name
Saving Songs	1st Input for Song Name

Exiting the Music Player

You may exit the Music Player by typing CTRL-C if you are being prompted for a song name; you are then returned to the Main Menu 1 or 2. After each composition plays, the Music Player automatically prompts you for another composition file name; CTRL-C at this point returns you to one of the Main Menus.

Chapter VIII
The Instrument Definer

With the Instrument Definer program you can create your own "instruments" to play the songs you enter with the Music Editor. Your instrument definitions can either be modeled on traditional instruments or can be your own unique creations. You can get just the sound you want by developing special instrument definitions for each song you want to play.

You can create an instrument by actually designing waveforms and then giving them amplitude and frequency histories. Logical oscillators are then assigned to interpret these waveforms as musical sounds. The Instrument Definer program helps you through each step, and before you know it you'll be designing a variety of instruments.

You don't have to understand how waveforms or oscillators work to use the Instrument Definer. In fact, you don't even have to know what they are. Instead, you can create an instrument "by ear". The Instrument Definer gives audio feedback during each stage of the design process so you can hear what an instrument sounds like as you create it. Simply change the instrument until its sound matches your expectation.

If you have never used the Instrument Definer program before, you should read the tutorial section of this chapter carefully, doing all the examples as they are presented. If you are somewhat familiar with the Instrument Definer and are looking for specific information, skip the tutorial section and go straight to the reference section. If you want more technical details, see Chapter 9, MusicSystem Theory.

What Is An Instrument Definition?

An Instrument Definition is made up of information which causes a group of logical oscillators (or a single oscillator) to generate pulsations which create a unique sound. Each logical oscillator contains a waveform, a frequency history and an amplitude history "envelope". The waveforms are created by a sub-program of the Instrument Definer, called the Wavemaker, and are made up of harmonics and their relative amplitudes. The envelope determines the overall amplitude history of the waveform for the duration of a single note. The frequency history determines the frequency of the waveform for the duration of a single note.

NOTE: The above explanation is very simplified. See the Instrument Definition portion of Chapter 9 for details of what makes up an instrument definition.

Tutorial Section

In this section you will find detailed directions for building an instrument definition. If you follow the directions outlined here, you will load an Organ instrument definition from among the instrument files supplied with the MusicSystem. You will then add to and modify the Organ to create an interesting electronic keyboard instrument. With the knowledge you gain from building this sample instrument, along with the information you'll find in the reference section of this chapter, you will be able to create an infinite number of varied instruments.

The System 2 Diskette

The Instrument Definer program is on the System 2 diskette. Put the System 2 diskette in your boot drive, and boot it with the procedures you usually use with your computer. (You'll find booting instructions in your Apple DOS manual.) In a few seconds the System 2 Main Menu will appear on the screen.

The System 2 Main Menu is very much like the System 1 Main Menu, but the options available from each are different. Notice that the Music Player is available from both Main Menus. You'll find that this is very convenient, as you will want to hear your newly completed instrument definition played in a song as soon as it has been created.

As with the System 1 Main Menu, some DOS commands can be typed from the System 2 Main Menu. To see what instrument files are available to start with, you can CATALOG your Instrument Files diskette. If you have one drive, put the Instrument Files diskette in your disk drive, and then type

    CATALOG

If you have two disk drives, put the Instrument Files diskette in your auxiliary drive, and type

CATALOG, Sn, Dn

where n is the slot and drive number, respectively, in which your auxiliary disk drive is plugged. A list of the files stored on the Instrument Files diskette will then appear on your screen.

Notice that these files all have either WAVE. or an IDEF. prefix. The files that are prefixed with IDEF. are instrument definition files. When you save an instrument definition, the IDEF. prefix will automatically be added to it. The files that are prefixed with WAVE. contain waveforms that are part of an instrument definition. Waveforms are fascinating. You'll get to create one as you follow along in this chapter. The Idef file called IDEF.ORGAN is the one you'll build on in this chapter.

Let's get started! (If you are using only one disk drive, reinsert the System 2 diskette.) Press the space bar to return to the System 2 Main Menu, as indicated at the bottom of the screen. Items on the System 2 Main Menu can be accessed with the keyboard, the light pen, or the game paddles. The Instrument Definer program, however, uses only keyboard input. With one of these methods choose the Instrument Definer from the System 2 Main Menu. In a moment the 14-option Instrument Definer Menu will appear on the screen.

Notice the line at the bottom of the screen that says

SELECT MENU NUMBER 1 TO 14
=>

The message on this line will change, depending on what you are expected to type. This line will be referred to in this section of the manual as the "prompt line". Whenever the Instrument Definer Menu, or one of the Instrument Definer program's sub-menus, is on the screen, the prompt line will indicate what you are expected to type.

Load an Instrument Definition

Before you can build an instrument definition with the Instrument Definer program you must have something to start with. One way to establish a base instrument that can be built on is to load such an instrument from the Instrument Files diskette. To load an instrument definition choose the first option from the Instrument Definer Menu, Load Instrument Definition, by typing a 1 and pressing the RETURN key.

When you load an instrument definition, any instrument definition that is currently being worked on will be replaced by the new instrument definition. For this reason the program gives you a chance to change your mind at the last minute by asking you to verify your choice.

VERIFY (Y/N) →

If you wanted to back out and cancel the load procedure, you could type an N for "no", and you would be prompted to choose another option. Since you really do want to load an instrument definition, type a Y for "yes". The prompt line will then say

IDEF FILE NAME
=>

When you see this prompt you can abort the load procedure by pressing the ESC key. For now, continue with the load procedure.

If you are using one disk drive, put the Instrument Files diskette into your disk drive, and type

ORGAN

and then press the RETURN key to load the instrument definition called IDEF.ORGAN. If you are using two disk drives, put the Instrument Files diskette into your auxiliary drive, type

ORGAN, Sn, Dn

and then press the RETURN key. Note that you don't type the IDEF. prefix when specifying a filename. The Instrument Definer program supplies the prefix for you.

If You Get an Error Message

If you mistyped the filename, or if the diskette containing the file is not in the disk drive you specified, you will get the message

DISK ERROR: FILE DOESN'T EXIST
FILE NAME: IDEF.ORGAN
INSERT IDEF DISKETTE INTO Sn, Dn
SPACE BAR WHEN READY
OR ESC TO CANCEL

If the wrong diskette is in the drive, insert the right diskette, and press the space bar. If you mistyped the filename, press the ESC key and re-choose the Load Instrument Definition option from the Instrument Definer Menu.

Audio Feedback

When the organ instrument definition has been successfully loaded, the Instrument Definer Menu will be repainted on the screen, but that's

not all that will happen. The organ you loaded will begin to generate audio feedback in the form of a C major scale. This scale will be played continuously, enabling you to hear the differences as you change the instrument definition.

Loud or Soft

To the right of option 9, Change Dynamic, on the Instrument Definition Menu you will see the value 100. This value represents the dynamic, or loudness, of the audio feedback. Every time you boot the Instrument Definer program the default dynamic value, 100, will be reinstated.

As you are working on instrument definitions you may find that softer audio feedback is desirable. (You can come up with some rather surprising noises.) To make the audio feedback softer, first choose option 9 from the Instrument Definer Menu. A new prompt will immediately appear:

DYNAMIC VALUE 0 TO 100
=>

For now let's reduce the audio feedback volume by half. Type the number 50 and press RETURN. The audio feedback will immediately become softer, and the prompt line will be reprinted on the screen. Try changing the audio feedback a few times.

Try giving a dynamic value of 0, and see what happens. If you wish to make the audio feedback louder than the greatest dynamic value, 100, will allow, use the volume control on your stereo amplifier.

Customized Audio Feedback

As you become familiar with the Instrument Definer, you will find it advantageous to create slightly different instrument versions for some songs. So that you can create instruments specially for certain songs and hear the results immediately, the Instrument Definer allows you to use a Play file as audio feedback.

Play files you produce with the Music Player can be used as audio feedback as long as they are not too large or complex. However, you will probably want to use a piece of music that has only one part. You can easily use the Music Editor to modify a multi-part Play file so that it has only the part for which you are developing an instrument definition.

In the general tutorial section of this manual (Chapter III) the steps for entering the musical notation for the well known song, "America", were given in detail. Let's use this song for now.

NOTE: Only Play files can be loaded with the Load Play File option. Comp files won't work. Use the Music Player to convert Comp files to Play files. (See Chapter VII, The Music Player for more information on Play files.)

Choose option 8, Load Play File, from the Instrument Definer Menu. The prompt line will display

PLAY FILE NAME
=>

Put the Song Files diskette that contains the Play file named America into your disk drive, your auxiliary drive if you are using two drives. Then type the name of the Play file you wish to load, in this case,

AMERICA

and press the RETURN key. In a few seconds American will begin to play. Replace your Song Files diskette with the Instrument Files diskette.

Change the audio feedback back to the major scale by choosing the Load Play File option and specifying DEFAULT when you are prompted for the Play filename.

Change the Pitch

Instrument Definitions sound dramatically different at different pitches. A brass instrument, for example, might sound like a tuba at a low pitch, a trombone at a slightly higher pitch, a trumpet at a higher pitch still, and just plain weird at a very high pitch. Taking this kind of variation into consideration, the Instrument Definer lets you change the pitch of the audio feedback with the Transpose option.

The audio feedback should now be playing the default song, a C major scale. Let's transpose the audio feedback so it will play an A major scale instead.

The Transpose option works like this. Each allowable transpose value represents one half step. The default transpose value that is assigned with the Instrument Definer program first comes on the screen is 0. The 0 doesn't represent a particular pitch value. Instead, it represents the original pitches of the notes in

the song being used as audio feedback. The pitch at transpose value 0 actually changes, depending on the note that is being played.

For example, when the default song is being used as audio feedback, 0 refers to the original pitches of each note of the scale. If you enter a Transpose value of 2, each note of the scale will be moved up 2 half steps. Thus to play a major scale that starts at the A below middle C, the transpose value must be changed to 3 half steps below middle C, or −3.

Choose option 10, Transpose, from the Instrument Definer Menu. The prompt line will display

TRANSPOSE VALUE −95 TO 95
=>

Type a −3 and press the RETURN key. The audio feedback will immediately change to an A major scale.

Transposing a real song, such as "America", works in the same way. To transpose "America", which is in the key of G, to the key of A, change the transpose value to 2. This will move the key up 2 half steps.

Delete an Oscillator

Now you are ready to begin the process of actually changing the Organ to an electronic sounding instrument. The organ is made up of two logical oscillators. The instrument you will create with the help of this chapter is also made up of two oscillators, but one of them is so different from those of the organ that it is more efficient to delete one oscillator and start over than to modify both existing oscillators.

Choose option 4, Delete Logical Oscillator, from the Instrument Definer Menu. You will then see the prompt

DELETE OSCILLATOR # 1 TO 2
=>

You need to delete oscillator number 2, so type a 2 and press the RETURN key. You will then be asked to

VERIFY (Y/N) →

If you changed your mind about deleting the oscillator, or if you mistyped the oscillator number, you could abort the deletion by typing an N in answer to this question. Since you do want to delete oscillator number 2, type a Y.

When you listen to the audio feedback, you will hear a difference. The organ sounds thin. That is because only one oscillator is now generating sound.

Add a New Oscillator

The next step is to add an oscillator to the instrument definition. Select option 3, Add Logical Oscillator, from the Instrument Definer Menu. In a few seconds an entirely different screen display will appear. This is the Logical Oscillator Menu. You can see that this menu is a little different from the choice menus you have used with the MusicSystem so far. Instead of choosing numbered options from this menu, you move the cursor to the option you wish to change or add. The prompt line works the same way it does from the Instrument Definer Menu. It will prompt you to type the needed information, and what you type will appear on the prompt line.

The left-pointing arrow at the right edge of the screen is a cursor. The cursor points to the current item about which information is needed. Right now the cursor is pointing to the wavename, ORGAN1. The waveform called ORGAN1 is left over from the organ instrument definition you are building on. If you hadn't loaded an instrument definition, the waveform supplied when you added a logical oscillator would be called DEFAULT.

Special control characters allow you to move the cursor from item to item. These are CTRL-W to move the cursor up and CTRL-S to move the cursor down. Notice that you can't move the cursor up past the wavename item or down past any unanswered items.

The wavename is the name of the waveform associated with the current logical oscillator. For now leave the Organ1 waveform in the oscillator. A little later you will create a completely new waveform and incorporate it into this oscillator.

Use CTRL-S to move the arrow cursor to the next item, Osc. Weight. The prompt line will say

OSC WEIGHT 1 TO 100
=>

The oscillator weight is the oscillator's importance or loudness relative to the other oscillators in the instrument definition. For the instrument you are creating you need two

oscillators of the same weight. You will give them both a weight of 100. Type

100

and press the RETURN key to give a weight of 100 to this oscillator. The cursor automatically moves to the next item, Exp. Decay Factor.

The Exponential Decay Factor determines the rate at which the sound of the instrument decays during the sustain portion of the envelope. (See Chapter IX for more information on the envelope.) For now, enter the value 500.

When you have entered the Exponential Decay Factor, the arrow cursor will move down one, and the prompt line will say

PRESS SPACE BAR
TO ENTER COORDINATES

The Amplitude History and the Frequency History both require that you enter groups of X and Y coordinates. A coordinate editor helps make coordinate entry more convenient. The coordinate editor is automatically invoked when you press the space bar. Do so now and the coordinate editor screen will appear.

Notice that the line of text at the top of the screen indicates that you are entering coordinates for the Amplitude History. This is because the arrow cursor on the Logical Oscillator Menu was pointing to Amplitude History when you went into the coordinate editor.

Every logical oscillator must have an amplitude history, and an amplitude history can have up to 15 sets of X and Y coordinates. The oscillator you are adding to the instrument definition has only 1 set of amplitude history coordinates. To enter them, type

100,100

and press the RETURN key. The coordinates you typed will appear near the top of the screen, and the arrow cursor will move down one space. The prompt line will now prompt you to type the next set of coordinates. Since you don't want to enter any more amplitude history coordinates, press the RETURN key again. The Logical Oscillator Menu will now appear on the screen.

The arrow cursor is now pointing to Frequency History. While the other items on the Logical Oscillator Menu are required for each logical oscillator, the frequency history is optional. The instrument you are creating does have one, however.

Press the space bar to enter the coordinate editor. Notice that the line of text at the top of the coordinate editor screen now indicates Frequency History.

As with amplitude history coordinates, there can be up to 15 sets of frequency history coordinates. The instrument you are creating, however, has only 1 set. Type

100,1

and press RETURN to enter the frequency history coordinates. When the prompt line prompts you for the next set of coordinates, press RETURN again to redisplay the Logical Oscillator Menu.

This oscillator is now complete. The next step is to replace its old waveform with a new one you will create. Press the RETURN key once more to return to the Instrument Definer Menu.

The Wavemaker

There is a special sub-program in the Instrument Definer that allows you to create waveforms for the logical oscillators in your instrument definitions. We'll use this program to create a waveform for the oscillator that was just added to the instrument definition.

Choose option 6, Create Waveform for Oscillator, from the Instrument Definer Menu. The prompt line will then ask

LOGICAL OSCILLATOR NUMBER 1 TO 2

Since you are going to create a waveform for oscillator number 2, type a 2 and press RETURN. After a moment's delay the Wavemaker Menu and Graph will appear on the screen.

The waveforms created with the Wavemaker sub-program consist of harmonics and their relative weights. The harmonics appear as lines on the Waveform Graph. The numbers at the bottom of the Waveform Graph represent the harmonic numbers, and the numbers at the left edge of the Graph represent the harmonic weights. Right now the old waveform in oscillator number 2 (called ORGAN1) is graphed. Clear the old waveform by selecting option 1 from the Wavemaker Menu. You will be asked to verify the New option, after which a blank Waveform Graph will be painted on the screen.

The waveform you are creating will have 8 harmonics of various weights. Here's a list of the harmonics you should include:

Harmonic	Weight
1	100
2	60
3	40
4	25
5	15
6	10
7	5
8	3

Now choose option 2, Change Harmonic, from the Wavemaker Menu. The prompt line will say

HARMONIC NUMBER 1 TO 24
=>

Type the first harmonic number on the above list, 1, and press RETURN. The prompt line will then say

WEIGHT 0 TO 100
=>

Type the weight that corresponds to the first harmonic on the above list, 100, and again press the RETURN key. A line representing the harmonic you just entered will immediately appear on the Wavemaker Graph.

The prompt line will now ask you for another harmonic number. Enter all the harmonics and their weights that appear on the above list. There will be a slight delay between each entry. Wait for the prompt line to be reprinted before you continue typing. When you have entered all 8 harmonics, press the RETURN key to indicate that you are through entering harmonics.

If you like, you can now look at the wave you entered. When the prompt line asks for another menu selection, choose option 4, Plot Waveform, to see the wave. The old screen will be erased, and the waveform you entered will appear.

When you have finished looking at the waveform, press the space bar as indicated at the bottom of the screen. This will cause the Wavemaker Menu and Waveform Graph to reappear.

To incorporate the newly created waveform in the current logical oscillator, choose option 5, Save and Exit, from the Wavemaker Menu. The prompt line will then display

EXIT
VERIFY (Y/N) →

Since you do want to exit the Wavemaker subprogram, type a Y. Don't panic! Your waveform is still safe. If you wait just a moment you will see the prompt line display

SAVE
VERIFY (Y/N) →

Type a Y to save the waveform. You will then be prompted to type a name for the new waveform.

WAVE FILE NAME
=>

Let's give this waveform the name "Pretty". Type

PRETTY

and press the RETURN key. In a few moments the new waveform, Pretty, will be incorporated into the current logical oscillator, oscillator number 2. The Instrument Definer Menu will then appear on the screen again.

Modify an Oscillator

The next step is to modify oscillator number 1 so that it makes a slightly different sound. When modifying an oscillator you use the Logical Oscillator Menu, just as you do with the Add Logical Oscillator option. Choose option 5, Modify Logical Oscillator, from the Instrument Definer Menu. The prompt line will display

MODIFY LOGICAL OSCILLATOR
NUMBER 1 TO 2
=>

Type a 1, since the oscillator you want to modify is number 1. The Logical Oscillator Menu will then appear on the screen. This time, however, all the items on the menu will be filled in with information.

The first thing to do is load a new waveform from the Instrument Files diskette. However, the arrow cursor is not pointing to the Wavename option now. Move the cursor to the item you wish to modify. Use CTRL-W to move the cursor up to the first item, Wavename.

When the prompt line says

WAVE FILE NAME
=>

insert the Instrument Files diskette in your disk drive, drive 2 if you are using more than one disk drive. A special waveform called UGLY has been included with the instrument definitions supplied with the MusicSystem.

This waveform is specifically for the instrument definition you are creating. Type the wave filename

UGLY

The Wavename on the menu will be changed from ORGAN1 to UGLY. Notice that the prompt line again asks you for a wave file name. When you are using the Modify Logical Oscillator option, you must indicate when you are finished modifying an item by moving the cursor with the CTRL-W and CTRL-S commands.

The oscillator weight doesn't need to be changed, so use the CTRL-S command to skip to the Exp. Decay Factor item. Enter an exponential decay factor of 1000 for oscillator number 1.

Use CTRL-S again to move the cursor down to Amplitude History. You will use the coordinate editor to enter the amplitude and frequency histories, just as you did when you added an oscillator.

Press the space bar to enter the amplitude history coordinates. You will enter two sets of coordinates this time. They are

5 , 100
100 , 40

When you enter these coordinates they will replace the old ones that appear on the screen. When the amplitude history coordinates have been entered, press the RETURN key to go back to the Logical Oscillator Menu.

Now move the cursor to the next item, Frequency History. Again, press the space bar to enter X and Y coordinates. You will enter one set of frequency history coordinates this time. They are

100 , -1

Enter these coordinates, and press the RETURN key to go back to the Logical Oscillator Menu.

You are now finished modifying oscillator number 1. If you wish to change something, use CTRL-W to move the cursor to the item you wish to change. When everything is to your liking, press the RETURN key from the Logical Oscillator Menu to go back to the Instrument Definer Menu.

Log vs. Linear

The attack history and decay for instrument definitions can be calculated either logarithmically or linearly. That probably doesn't mean much to you right now and, in fact, you don't need to know about logarithms to use this feature. Instead, listen to the audio feedback to tell which method you prefer for a particular instrument definition.

Option 11 on the Instrument Definer Menu, Log vs. Linear, lets you choose the method of calculation you want. When you choose this option the prompt line will display

LOG=0 LIN=1 0 TO 1
=>

The instrument definition you are working on now requires this option to be left at the Linear setting. Change the setting back and forth a few times to hear the difference. When you are through, reset this option to Linear by typing a 1.

NOTE: If you want more information on Log vs. Linear, see Chapter IX, MusicSystem Theory.

Attack Time

You are now ready to set the attack time for the entire instrument definition. You will find that the attack time setting can drastically affect the sound of the instrument.

Choose option 12, Select Attack Time, from the Instrument Definer Menu. The prompt line will look like this:

ATTACK TIME (EVEN NO.) 2 TO 124
=>

The instrument you are creating requires an attack time of 20, so type a 20 and press the RETURN key. Notice the change in the audio feedback.

Decay Time

Next you must set the decay time for the instrument definition. Choose option 13, Select Decay Time, from the Instrument Definer Menu. This option also requires an even integer.

DECAY TIME (EVEN NO.) 2 TO 124
=>

The instrument you are working on requires a very short decay time. Type a 6 and press the RETURN key. Again, take note of the difference in the audio feedback.

Display the Instrument

The Instrument Definer has a special feature that allows you to get a visual picture of the instrument you are defining. You can use this feature to look at the instrument you just created.

Choose option 7, Display Instrument Information, from the Instrument Definer Menu. In a moment the five-option Display Options Menu will appear on the screen.

The Display Statistics option lets you see the statistics for the current instrument definition and each logical oscillator in the current instrument definition. Choose that option now by typing a 1 and pressing the RETURN key. Immediately a list of information will appear on the screen.

The two blocks of information at the top of the screen are pertinent to the entire instrument definition. Below the statistics for the instrument definition as a whole you'll find information on the first of the oscillators that make up the instrument definition.

To see the statistics for the other oscillator, scroll down with CTRL-S. You can use CTRL-S and CTRL-W to change the displayed statistics. When you scroll past the available statistics, the Display Options Menu will return to the screen.

The Plot Amplitude History option lets you see what the amplitude history coordinates you entered earlier actually look like. You can plot the amplitude history of each of the oscillators that make up the instrument definition. When you've finished looking at an amplitude history, press the space bar to return to the Display Options Menu.

Plot Frequency History works much the same as Plot Amplitude History. The resulting graphs, however, are quite different.

Perhaps the most interesting of the display options is the Plot Waveform option. Choose option 4, Plot Waveform, from the Display Options Menu. Try looking at the waveforms for both oscillators. The waveform you created with the Wavemaker program looks like a wave, but the waveform you loaded from diskette can hardly be described as a wave. This waveform was not obtained through the Wavemaker program. (For more information on waveforms not created with the Wavemaker, see Chapter IX.)

When you have finished looking at your handiwork, choose option 5, Exit to Main Menu. This option doesn't take you to the System 2 Main Menu, but to the Instrument Definer Menu.

Save Your Instrument

Now that you have worked to create a valuable instrument definition, you will probably want to save it on your Instrument Files diskette so you can use it later.

To save the instrument, choose option 2, Save Instrument Definition, from the Instrument Definer Menu. The prompt line will say

IDEF FILE NAME
=>

You've probably never heard anything quite like this instrument definition. We'll give it a name you've probably never heard before either. Type the name

BLEEDLE

The name restrictions are in agreement with the standard DOS filename conventions. There are only two exceptions to the standard DOS conventions. One exception is that instrument definition filenames must not have more than 11 characters. The other exception is that spaces should not be embedded in filenames or you may run into trouble later with the Music Editor.

After you have typed the filename, the instrument definition will be saved on a diskette in your Files drive. If you are using one disk drive, the instrument will be saved on the diskette in your boot drive. If you are using more than one disk drive, the instrument will be saved on the diskette in your second drive.

Leaving the Instrument Definer

Now that this new instrument is saved on a diskette you can use it later to play songs that you develop with the other parts of the MusicSystem. To leave the Instrument Definer, choose the last option on the Instrument Definer Menu, Quit. If you were to use the Quit option before you had saved your instrument definition, the instrument would be lost. Be-

cause the consequences can be dire, you must verify the Quit option.

VERIFY (Y/N) →

If you type an N, the Quit command will be aborted and the current instrument definition will be left intact. To reboot one of the System diskettes, insert a System diskette into your boot drive and type a Y.

Playing the New Instrument

You should now have the System 2 Main Menu displayed on your screen. You may have noticed that there is a Music Player on this diskette, as well as the Instrument Definer. Let's use the Music Player to listen to your newly completed instrument playing a real song.

Before invoking the Music Player, do a catalog of your Song Files diskette so you can see what songs you can choose from. Put your Song Files diskette in your disk drive, or, if you are using two disk drives, in your auxiliary drive. Then, if you are using just one disk drive, type

CATALOG

If you are using two disk drives, type

CATALOG, Sn, Dn

where n is the slot and drive numbers, respectively, of your auxiliary drive.

Press the space bar as indicated on the prompt line, and choose the Music Player from the Main Menu. If you don't know how to use the Music Player, see Chapter VII, which discusses it in detail. You can use your new instrument definition just like the other instruments on the diskette.

Reference Section

The Instrument Definer program allows you to create your own instruments for use with the MusicSystem. On the next page is a simplified diagram of the program structure. You can refer to this diagram as you read this chapter.

Boot the System 2 diskette and choose item 2, Instrument Definer, from the Main Menu. The message

** WAIT — LOADING **

will appear on the screen, and shortly the Instrument Definer Menu will be displayed.

The Instrument Definer Menu

Fourteen choices are offered on the Instrument Definer Menu. To choose an option, type the number to the left of it, and press the RETURN key. The Instrument Definer Menu looks like the list on the next page.

INSTRUMENT DEFINER

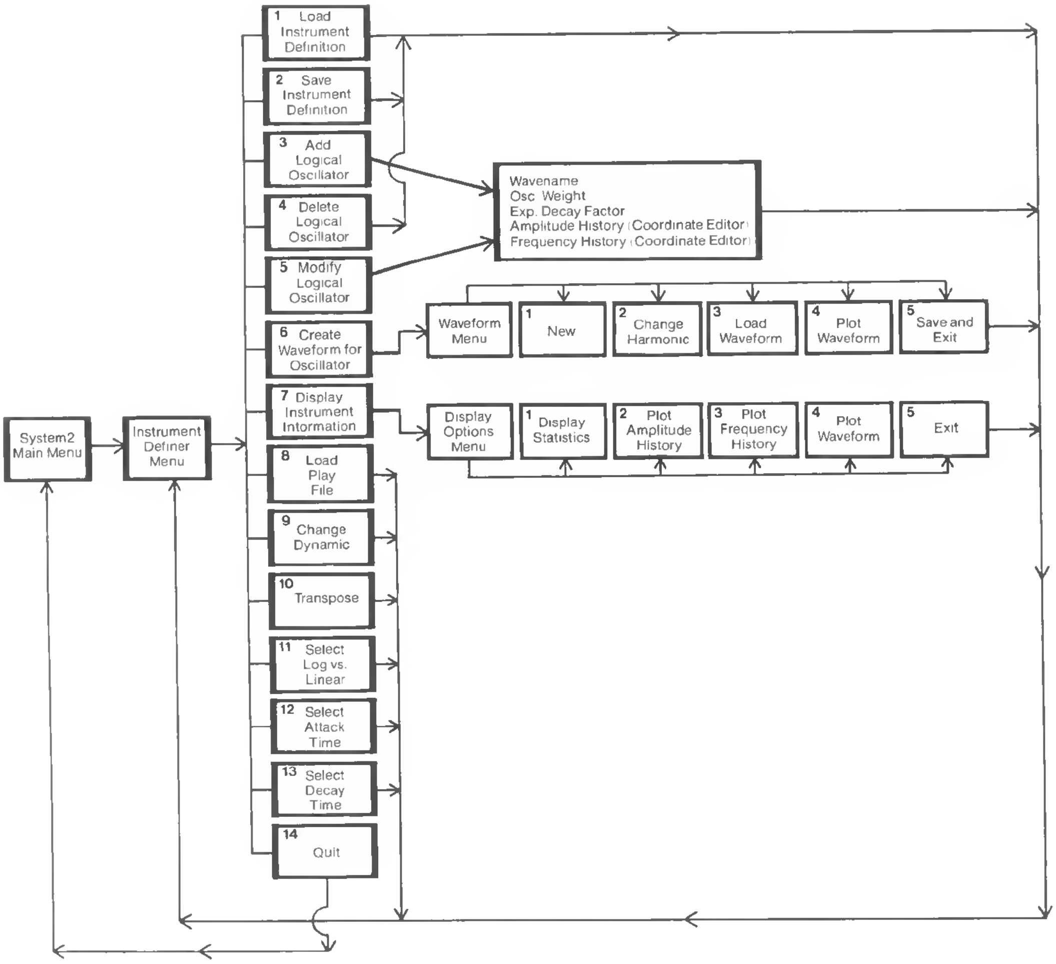
1. LOAD INSTRUMENT DEFINITION
2. SAVE INSTRUMENT DEFINITION
3. ADD LOGICAL OSCILLATOR
4. DELETE LOGICAL OSCILLATOR
5. MODIFY LOGICAL OSCILLATOR
6. CREATE WAVEFORM FOR OSCILLATOR
7. DISPLAY INSTRUMENT INFORMATION
8. LOAD PLAY FILE
9. CHANGE DYNAMIC 100
10. TRANSPOSE 0
11. SELECT LOG. VS. LINEAR LINEAR
12. SELECT ATTACK TIME 12
13. SELECT DECAY TIME 10
14. QUIT

SELECT MENU NUMBER 1 TO 14
=>

The information that appears to the right of menu options 9 through 13 will change depending on the current instrument definition (the one that is loaded). The ones displayed when you first invoke the Instrument Definer are the default values.

NOTE: The keyboard is the only input medium for the Instrument Definer; the light pen and game paddles cannot be used. A line editor has been provided to make keyboard input easier. Whenever you are at an input line (when the prompt line prompts you to type something), you can use the line editor functions. Use CTRL-X to erase the entire input line, or use the left-pointing arrow key to erase the line character by character.

Following is a description of each item on the Instrument Definer Menu.



Loading Instrument Definitions

This option lets you load an existing instrument definition from diskette. To load an instrument definition choose the first option from the Instrument Definer Menu. If there is already an instrument definition in memory, loading a new instrument definition will replace the old one. For this reason you must verify your choice before continuing.

VERIFY (Y/N) →

If you answer with an N, the Load process will be aborted. If you answer with a Y you will be prompted to type the IDEF, or instrument definition filename.

IDEF FILE NAME
=>

At this point you can press the ESC key to abort the Load process. Or, to continue the Load procedure, type the name of the instrument definition you wish to load. Do not include the IDEF. prefix that appears before the instrument name in the diskette catalog.

If you are using two disk drives, you can use the boot drive for the System diskettes and specify the second drive as the "Files" drive in which you will put your Instrument Files and Song Files diskettes. To do this, specify the slot and drive numbers of your second drive the first time you give the name of the instrument definition to be loaded. For example, to load the organ instrument definition, if your boot drive is drive 1 and both drives are plugged into the same disk controller card, type

ORGAN, D2

After this you won't have to specify the slot and drive numbers. The Instrument Definer

will assume the second drive is the Files drive until you indicate otherwise or re-boot the system.

If you are using one disk drive, you will need to switch between the System 2 and the Instrument Files diskettes. The Instrument Definition program will prompt you when you need to make the switch with this message:

INSERT IDEF DISKETTE INTO Sn, Dn
PRESS SPACE BAR WHEN READY

You will then switch diskettes and press the space bar.

These are the standard MusicSystem disk conventions, and they hold true throughout this chapter. This chapter will assume that you are familiar with the method you are using.

If the instrument definition does not exist on the diskette you specified, you will get the "File Not Found" message:

DISK ERROR: FILE DOESN'T EXIST
FILE NAME: IDEF.ORGAN
INSERT IDEF DISKETTE INTO Sn, Dn
SPACE BAR WHEN READY
OR ESC TO CANCEL

At this point you have two options:

- You can search for the instrument definition by inserting Instrument Files diskettes into the indicated drive, one at a time, pressing the space bar after each one. The Instrument Definer program will search the diskettes for the instrument definition. The error message will repeat each time the file is not found.
- You can press the ESC key to cancel the load command, in which case you will be prompted to choose from the Instrument Definer Menu again.

When the instrument has been successfully loaded, the Instrument Definer Menu will be re-painted on the screen, and the instrument will begin to give audio feedback in the form of a scale that is played over and over continuously. This scale is known as the "default song". (The default song can be changed to a real song. See the section in this chapter called Loading a Play File for details.)

Saving Instrument Definitions

With this option you can save the current instrument definition (the one that has been loaded or created) to diskette. Choose option 2 from the Instrument Definer Menu. You will be prompted

IDEF FILE NAME
=>

You can press the ESC key from the filename prompt to cancel the save operation. To continue the save process, type the name under which you wish to save the current instrument definition. The name you type must not be longer than 11 characters, and should not have embedded spaces. If you embed spaces in the name, you may run into trouble later when you are using the Music Editor.

A prefix, IDEF., will be added to your instrument definition name by the Instrument Definer program. This prefix identifies the file to the Instrument Definer program as an instrument definition.

When the instrument definition has been successfully saved, the audio feedback will resume.

Adding a Logical Oscillator

An instrument definition can have up to eight logical oscillators. To keep track of all these oscillators the Instrument Definer program numbers them, starting with 1. Each time you add an oscillator to your instrument definition the next highest number will be assigned to that oscillator. For example, if you add a third oscillator to an instrument definition that already has two, the new oscillator will be assigned the number 3.

NOTE: If you add an oscillator to an existing instrument definition, the waveform from the preceding oscillator will be incorporated into the new oscillator until you create or load a new waveform.

To choose option 3, Add a Logical Oscillator, from the Instrument Definer Menu, type 3 and press the RETURN key. A sub-menu will immediately appear on the screen. This submenu is called the Logical Oscillator Menu:

ADD LOGICAL OSCILLATOR #1

WAVENAME DEFAULT ←
OSC. WEIGHT
EXP. DECAY FACTOR
AMPLITUDE HISTORY
FREQUENCY HISTORY

WAVE FILE NAME
=>

This menu is different from other menus in that the five items on it are not really options at all.

The five-item list at the left of the screen is a list of oscillator parameters. For each oscillator you wish to add you must supply values for the parameters on the list. The information you enter will appear to the right of the corresponding parameter. Information must be given for each parameter except the last one, Frequency History, for the oscillator to be complete. Frequency History is optional.

You can use the ESC key to cancel the Add Logical Oscillator command when the prompt line at the bottom of the screen asks you to type the WAVE FILE NAME, OSC. WEIGHT, EXP. DECAY FACTOR, or ENTER COORD. If you use the ESC key, you will be returned to the Instrument Definer Menu, and the logical oscillator you were working on will be lost.

You can move the cursor from question to question and change your answers if you wish. Two control characters, CTRL-W and CTRL-S, control the cursor. CTRL-W moves the cursor up, and CTRL-S moves it down. These control characters will only move the cursor between those questions that have been answered. You can't proceed past a question that has not been answered yet.

Following is a brief description of each parameter.

Wavename

The word, Default, that appears next to the Wavename parameter represents the default waveform that will be used if you don't specify one and there are no other logical oscillators loaded. (The default waveform is the same as the Organ1 waveform.) The wavename you type will replace the Default.

In answer to this question, type the name of the waveform you wish to use for the new oscillator. Any waveform that exists on diskette can be used. (You can create your own waveforms with option 6, Create Waveform, discussed later in this chapter.) The waveform you specify will then be loaded from diskette. If you have indicated your second drive as the Files diskette drive, the Instrument Definer will look for the waveform on the diskette in that drive.

If the waveform is not on the diskette in your Files drive, you will get the File Not Found message. If you get this message, you can insert your other files diskettes one at a time and search each of them, in turn, for the waveform file.

When the waveform has been successfully loaded, the wavename will appear on the screen to the right of the WAVENAME prompt, and the cursor will point to the next question on the list. The prompt at the bottom of the screen will ask you for the oscillator weight.

Osc. Weight

The oscillator weight determines the loudness of an oscillator in relation to the other oscillators that make up the instrument definition. For example, let's say you have a two oscillator instrument definition and you want oscillator 1 to be twice as loud as the oscillator 2. You could give oscillator 1 a weight of 100 and oscillator 2 a weight of 50. Oscillator weights are relative, so you could get the same effect by giving oscillator 1 a weight of 50 and oscillator 2 a weight of 25. The Oscillator Weight parameter must be given a value that is an integer from 1 to 100. When you have given an acceptable oscillator weight, the cursor will move down to the next question.

Exp. Decay Factor

The Exp. Decay Factor, or exponential decay factor, determines how fast or slow the sound of the new oscillator will decay or fade during the sustain portion of a note. This parameter must be supplied with an integer from 0 to 2048.

When the question has been answered with an acceptable answer, the cursor will move down to the next question, and the prompt at the bottom of the screen will ask you to

PRESS SPACE BAR
TO ENTER COORDINATES

Amplitude History and Frequency History or, the Coordinate Editor

The last two questions on the Oscillator Menu require that you enter sets of X and Y coordinates. A Coordinate Editor has been provided to make this process easier. To invoke the Coordinate Editor, press the space bar as instructed by the prompt line.

As soon as you invoke the coordinator editor, the screen display will change. The top line will indicate which oscillator you are working on and whether you are entering Amplitude History or Frequency History coordinates. An arrow cursor will appear on the right edge of the screen, indicating where the coordinates you enter will appear. At the bottom of the screen you will be prompted to

INPUT COORD X(1 TO 255),
Y(−127 TO 127)
=>

Coordinates are entered as two integers separated by a comma, with the X coordinate first. You may enter from 1 to 15 coordinate pairs. The coordinates will appear on the screen, separated by commas. The cursor at the right edge of the screen will scroll down one line each time you enter a set of coordinates.

If you notice that you've made a typing error, you can use CTRL-X to erase the line you just typed if it hasn't been entered yet. If you don't want to erase the entire line, you can use the left-pointing arrow key to backspace one at a time over the characters you wish to erase.

All X coordinates must be given in ascending order. For example, you can't give 24 as the first X coordinate and 13 as the second X coordinate. If you give coordinates that are out of range or out of order, you will get an error message and be allowed to try again.

If you wish to change some of the coordinates you have entered, you can move the cursor from row to row with CTRL-W and CTRL-S. CTRL-W moves the cursor up and CTRL-S moves it down.

NOTE: If a control character is used, it must be the first character in the buffer. To insure that control characters are executed, always type them as the first characters on a line. CTRL-X is an exception to this rule.

To delete a pair of coordinates that has already been entered, move the cursor to the row of the coordinates you wish to remove, then press CTRL-D. Any coordinate pairs displayed below the deleted pair will be moved up one row, filling in the extra space created by the deletion.

The CTRL-D command must be the first character in the line or it will not be recognized. For example, you can't start to type a coordinate pair, change your mind in the middle of the entry, and then press CTRL-D.

To insert a set of coordinates between two existing coordinate pairs, move the cursor to the place directly above where you wish the new pair to appear. Then press CTRL-I and type the new coordinates. The coordinates below the new pair will be moved down one row to make room. As with other control characters, CTRL-I must be the first character in the buffer.

To replace a pair of coordinates, move the cursor to the pair you wish to replace. Then simply type the new coordinates. The new set will be substituted for the old.

From the Coordinate Editor you can use the ESC key to abort the Add Logical Oscillator function. If you press the ESC key from the Coordinate Editor, the Instrument Definer Menu will appear on the screen, and the logical oscillator you are working on will be lost.

When you have entered all the Amplitude History coordinates you want, press the RETURN key in answer to a prompt to go back to the Logical Oscillator Menu. The arrow cursor will point to the next item, Frequency History, and the newly added oscillator will begin to generate audio feedback.

Enter the Frequency History coordinates in the same way as the Amplitude History. Then, when you have finished entering coordinates, press the RETURN key to return to the Logical Oscillator Menu again.

NOTE: It is not necessary to enter Frequency History coordinates to complete an oscillator. Frequency History coordinates are optional. However, all the other parameters must be supplied to complete the oscillator.

When you are satisfied with your inputs, press the RETURN key to return to the Instrument Definition Menu.

Deleting a Logical Oscillator

This option lets you delete an oscillator from the current instrument definition. To choose this option, type 4 and press RETURN. The prompt at the bottom of the screen will say

DELETE OSCILLATOR # 1 to n →

where n is the number of oscillators currently being used.

If you wish to abort the command, press the ESC key. To continue the command, type the number of the oscillator you wish to delete. If you try to type a number that is out of range, the Apple will beep and wait for you to type an acceptable number. When the number has been accepted you will be asked to verify the command. Answering N will abort the com-

mand. Answering Y will complete the deletion. When an oscillator has been removed, the remaining oscillators are renumbered, if necessary, to reflect the new oscillator numbers.

Modifying a Logical Oscillator

This option allows you to change an existing oscillator. Choose option 5 from the Instrument Definer Menu. If the current instrument definition has more than one oscillator, you will be prompted

MODIFY LOGICAL OSCILLATOR
NUMBER 1 TO n
=>

(The n refers to the number of oscillators in the instrument definition.)

When you have entered the logical oscillator number, the Logical Oscillator Menu will appear on the screen. This is the same menu you used to Add a Logical Oscillator. To modify the oscillator, change the answers to the questions on the menu, using CTRL-W and CTRL-S to move among items. The changes you make will be immediately reflected in the audio feedback generated by the oscillator in question.

To cancel the modification, press the ESC key when the prompt line asks you for a parameter. To incorporate the modified logical oscillator into the current instrument definition, press the RETURN key when the prompt line asks you for a parameter.

Creating a Waveform

Option 6, Create Waveform, invokes the Wavemaker sub-program. With the Wavemaker section of the Instrument Definer program you can create a new waveform, or modify an existing waveform to your specifications. Choose option 6, Create Waveform, from the Instrument Definer Menu. The Wavemaker sub-program will then be loaded from the System 2 diskette.

If there are no logical oscillators currently in memory, you will get the message

NO LOGICAL OSCILLATORS

PRESS SPACE BAR WHEN READY

Pressing the space bar will abort the Wavemaker program and return the Instrument Definer Menu to the screen.

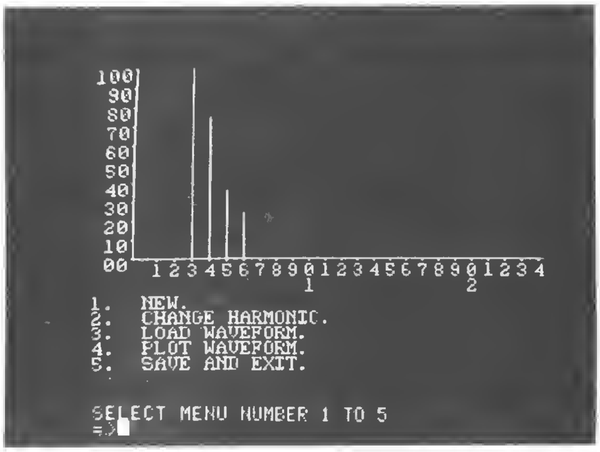
If there is only one oscillator in the current instrument definition, the Wavemaker Menu and Graph will automatically be displayed on the screen.

If there is more than one oscillator in the current instrument definition, you will be prompted

LOGICAL OSCILLATOR NUMBER 1 TO n
=>

(Again, the n in this statement refers to the number of oscillators in the current instrument definition.) Type the number of the logical oscillator for which you wish to create a new waveform. After a short delay, another submenu, the Wavemaker Menu, will appear on the screen along with the Wavemaker Graph.

The Wavemaker Graph indicates the harmonics that make up the waveform. The Wavemaker Menu lists the options available in the Wavemaker. The screen display looks like this:



NOTE FOR ADVANCED PROGRAMMERS: It is possible to create your own program for generating waveforms. However, waveforms created in this way are slightly different from those created with the Wavemaker. Waveforms created with your own program can be loaded into the Instrument Definer and used within a logical oscillator, but these waveforms will not be recognized by the Wavemaker program. If you load one of these waveforms into the Wavemaker, all the harmonics will be set to 0, and nothing will appear on the harmonics graph. For more information on creating waveforms, see Chapter IX, Music-System Theory.

The numbers on the left edge of the graph represent the harmonic weights in percentages.

The numbers at the bottom of the graph indicate the harmonic that is being graphed. The vertical lines on the Wavemaker Graph represent the weight percentages of the harmonics for the waveform you specified.

Following is a description of the five options available from the Wavemaker Menu.

New
Choosing this option from the menu causes the graph display to be cleared and the current waveform to be disregarded so you can build another waveform. Since the current waveform is destroyed by this process, the program asks you to verify that you indeed wish to destroy the waveform.

VERIFY (Y/N) →

When you have verified your choice the screen will be re-painted, the harmonics graph will be cleared, and the audio feedback will change.

Change Harmonic
The Wavemaker can plot and incorporate into a waveform up to the twenty-fourth harmonic. When you choose the Change Harmonic option, you will be asked to give the harmonic number:

HARMONIC NUMBER 1 TO 24
=>

After you have typed the harmonic number, you will be prompted to type the percentage weight you wish to be given that harmonic:

WEIGHT 0 TO 100
=>

When you are prompted for either the harmonic number or the harmonic weight, you can press the ESC key to abort the Change Harmonic option. If you press ESC, you will again be prompted to select a menu option.

As soon as you have entered these items, the audio feedback will change, dramatically if you give the harmonic a large weight percentage, and less if you give the harmonic a small weight percentage. If you are working with an instrument definition that has many logical oscillators, the effect of the change in harmonic may be masked by the other waveforms. You will be asked for new harmonics repeatedly until you press the RETURN key instead of typing a parameter.

The harmonics that make up waveforms are divided into six groups, with harmonics 1 through 4 in the first group, harmonics 5 through 8 in the second group, and so on. Each of these groups is stored on the System 2 diskette, each in a special file. Each time you enter harmonics in a different group, the System 2 diskette is accessed. To save time and avoid unnecessary disk access enter harmonic information for one group at a time. For example, if you want to use harmonics 1, 2, 3, 5, and 6, enter the information for harmonics 1, 2 and 3 first, then 5 and 6.

When you have finished entering harmonics, press the RETURN key, and you will be prompted to choose another option from the Wavemaker Menu.

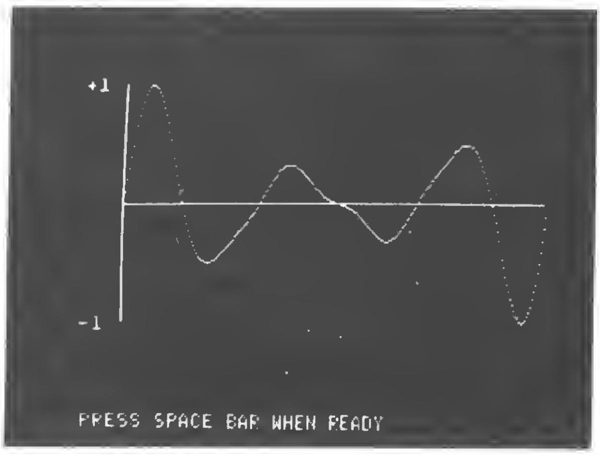
Load Waveform
This option allows you to load an existing waveform from diskette. When you choose this option you will be prompted

WAVE FILE NAME
=>

At this point you can abort the load procedure by pressing the ESC key. To continue loading, type the name of the waveform you wish to load.

The waveform will take some time to load. When the waveform is successfully loaded, its harmonics will appear on the graph and the oscillator associated with it will alter the audio feedback.

Plot Waveform
Choosing this option from the Wavemaker Menu will cause the screen to be cleared and the current waveform to be plotted on the high-resolution graphics screen. A waveform must be loaded or a blank graph will be displayed. The illustration below shows a sample waveform.



When you've finished looking at the waveform, press the space bar as indicated at the bottom of the screen to return to the Wavemaker Menu.

Saving and Exiting

With this option you can save a waveform on diskette and exit the Wavemaker, or you can exit the Wavemaker without saving the current waveform. When you save a waveform, it is automatically incorporated into the current instrument definition and saved on the diskette in the Files drive.

NOTE: Saving an instrument definition does not save the waveform associated with it! You must save the waveform separately.

To save a waveform or just exit the Wavemaker, choose the Save and Exit option from the Wavemaker Sub-Menu. You will then be prompted

EXIT
VERIFY (Y/N) →

If you answer with an N, you will be prompted to choose another menu item. If you answer with a Y, you will be prompted

SAVE
VERIFY (Y/N) →

Use caution when using this command. If you press the N key, the current waveform will be destroyed, and the Instrument Definer Menu will appear on the screen.

If you answer with Y, you will be prompted for the waveform filename:

WAVE FILE NAME
=>

The name you type must not have more than 11 characters. Except for the 11-character limit, the standard DOS filename conventions apply here.

NOTE: If you have a waveform which is associated with more than one logical oscillator in your instrument definition, and you save a new waveform under the same name, the internal waveform data for the other oscillator (or oscillators) will continue to reflect the old waveform data. To remedy this, either save and reload the entire instrument definition, or choose option 5, Modify Logical Oscillator, for the other oscillator, and retype the wavename to load the new waveform.

When you have entered a waveform name, the current waveform will be saved on the diskette in the Files drive, and the Instrument Definer Menu will return to the screen.

Displaying Instrument Information

This option allows you to display on the screen one of several pieces of information about the current instrument definition. These include oscillator statistics, amplitude and frequency histories, and waveforms.

When you choose this option from the Instrument Definer Menu a sub-menu, the Display Menu, will appear on the screen. The Display Menu looks like this:

DISPLAY OPTIONS

1. DISPLAY STATISTICS
2. PLOT AMPLITUDE HISTORY
3. PLOT FREQUENCY HISTORY
4. PLOT WAVEFORM
5. EXIT TO MAIN MENU

SELECT MENU NUMBER 1 TO 5
=>

These five options let you see the condition of the current instrument definition. The display options will not affect the instrument definition; they are solely for your information.

Display Statistics

The Display Statistics option lists the following information for the current instrument definition.

- Number of logical oscillators
- Current attack, decay, dynamic and transpose values
- Current audio feedback song name

In addition, the following information is printed for each logical oscillator.

- Oscillator number
- Oscillator weight
- Exponential decay factor
- Waveform name
- Amplitude history coordinates
- Frequency history coordinates

Only one logical oscillator's information is printed on the screen at a time. You can scroll the information up and down to see the information for the other oscillators. Use CTRL-W to scroll up and CTRL-S to scroll down. If you scroll past the information on the screen with CTRL-W or CTRL-S, the Display Menu will

return to the screen. Pressing the ESC key will return the Instrument Definer Menu to the screen.

Plot Amplitude History
This option plots the Amplitude History coordinates for one of the oscillators in the current instrument definition. The resulting graph shows you a visual picture of the relationship among the Amplitude History coordinates.

When you choose the Plot Amplitude History option from the Display Menu, you will be prompted

PLOT AMPLITUDE:
OSCILLATOR # 1 TO 3
=>

(The numbers on this prompt will vary, depending on the number of oscillators in the current instrument definition.) Type the number of the oscillator whose amplitude you wish to plot.

When the logical oscillator number has been accepted, the graph will appear on the screen. Near the bottom of the screen the following message will appear:

PLOT AMPLITUDE: OSCILLATOR #n

PRESS SPACE BAR WHEN READY

(where n is the number of the current oscillator). When you have finished looking at the graph, press the space bar to return the Display Menu to the screen.

Plot Frequency History
Plotting the frequency history is similar to plotting the amplitude history. When you choose this option, you must supply the oscillator number.

The Frequency History graph itself, however, looks somewhat different, as you will see. The Y axis values can be from –127 to 127, with zero corresponding to the middle point.

If the oscillator you specified has no frequency history, the message

NO FREQUENCY POINTS

PRESS SPACE BAR WHEN READY

will appear on the screen. If the oscillator does have a frequency history, the Frequency History graph will appear on the screen. As with the Amplitude History plot option, you can press the space bar to return to the Display Options Menu.

Plot Waveform
When you choose this option, you will again be prompted to type the oscillator number (if there is more than one oscillator in the current instrument definition). The same range conditions that affect the Amplitude and Frequency History plot options also apply to the Plot Waveform option.

The Waveform graph is plotted on the high-resolution screen. Press the space bar to return to the Display Menu.

Exit
NOTE: The Exit option does not take you to the System 2 Main Menu as suggested on the Wavemaker Menu. Instead, it leaves the Display Options Menu and returns the Instrument Definer Menu to the screen. Type 5 and press RETURN, to display the Instrument Definer Menu.

Loading a Play File

The default song that plays while you are defining an instrument is a simple scale. While the scale is adequate in many instances, if you want to create special instruments for a particular song, you may wish to make that song, or part of it, the Instrument Definer audio feedback song. The Load Play File option allows you to do this easily.

To load a Play file, choose option 8 from the Instrument Definer Menu. The prompt line at the bottom of the screen will say

PLAY FILE NAME
=>

If you wish to cancel the command, press ESC at this point. To continue the command, type the name of the Play file you wish to load, and press the RETURN key. (Make sure the diskette in the Files drive has the Play file on it.)

There is a limit to the size of Play file that can be loaded into the Instrument Definer. The allowable size is determined by the length and complexity of the song. The number of parts, the polyphony of each part, and other factors determine the complexity of the song. Because so many factors are involved, we can't tell you the exact size limit. To find out if a song will fit, try it. If the song is too large, you will get an error message.

Up to eight logical oscillators can be used at one time, and up to four logical oscillators can

sound from a stereo speaker at once. You may have already discovered that, when using the Music Player, you can accidentally go over the allowed number of oscillators for one speaker by combining instruments that have too many logical oscillators. When you use the Load a Play File option from the Instrument Definer Menu, all the parts in the song will be played with the current instrument definition. If there are several parts to the song, or if the current instrument definition has several parts, the allowed number of logical oscillators for one speaker may be exceeded. If this happens, you will get a message to this effect. Since it is usually better to use a simple one-part song for audio feedback, you probably won't run into this problem very often.

When you load a Play file into the Instrument Definer, all the parts will be played by the current instrument in the Instrument Definer. If the Play file has several parts or the current instrument has several oscillators, you may have too many instruments on one speaker.

If the Play file was not found, you will get the File Not Found message. Put another Files diskette into your Files drive and press the space bar to search for the file on that diskette.

If the load is unsuccessful, the default song, the scale, will be reinstated. When the Play file is loaded successfully, the song will begin to play.

Changing Dynamic

This option allows you to change the volume of the audio feedback. The default dynamic value is 100, the maximum volume. You may want to lower the volume while you are working on the instrument.

To change the dynamic value, choose option 9, Change Dynamic, from the Instrument Definer Menu. The following prompt will then be displayed:

DYNAMIC VALUE 0 TO 100
=>

If you change your mind and wish to cancel the command, press the ESC key at this point.

A value of zero will give the least volume (actually, no volume), and a value of 100 will give the greatest volume. Type an integer from 0 to 100 to represent the dynamic value.

If an instrument definition is loaded, the audio feedback will immediately reflect the change in dynamic, and the value you entered will appear on the Instrument Definer Menu screen to the right of the Change Dynamic option.

Transpose

You will find that the instruments you create will often sound very different at different pitches. The Transpose option lets you change the pitch of the audio feedback to hear what an instrument sounds like at different pitches.

When you choose option 10, Transpose, from the Instrument Definer Menu you will be prompted

TRANSPOSE VALUE –95 TO 95
=>

To abort the command, press the ESC key. To continue the command, type an integer in the range specified. The number you type represents the number of half-steps the starting pitch is raised or lowered. For example, to change the pitch one octave, give a transpose value of 12 or –12. Lower numbers will lower the pitch and higher numbers will raise it.

When the Transpose value has been accepted, it will appear on the Instrument Definer Menu screen to the right of the Transpose option, and the audio feedback will begin to play at the new pitch.

NOTE: The Transpose option allows you to enter 191 different transpose values (–95 to 95). The MusicSystem can generate only 96 different pitches, numbered from 0 to 95. This may seem strange to you until you realize that transpose values are relative, whereas absolute pitch values stay the same. For example, let's say you have an audio feedback "song" comprised of four notes of the same pitch, and this pitch happens to be 95, or the highest pitch the MusicSystem can make. Let's also say that you wish to transpose the song to pitch 0, the lowest pitch the MusicSystem can make. Now, here's where it gets somewhat complicated. The initial pitch of the song, 95, is given transpose value 0. This means that pitch 95 is, in this case, the same as transpose value 0. In order to change pitch 95 (transpose value 0) to pitch 0, you would have to give a transpose value of –95. The moral of this story is absolute pitch values and transpose values are not the same at all. And for good reason!

Select Log vs. Linear

This option lets you choose whether the decay curve and the amplitude history for the attack will be specified logarithmically or linearly. You don't have to understand logarithms to use this option. Just try both settings, listening to the audio feedback to check the results.

When you choose this option from the menu, the prompt line will display

LOG=0 LIN=1 0 TO 1
=>

You can abort the command at this point by pressing the ESC key. To continue with the command, type either a 0 or a 1.

When the value has been accepted, the change will be indicated at the right of the option on the menu and will immediately become apparent in the audio feedback.

If you want more information on Log vs. Linear, see Chapter IX.

Select Attack Time

This option lets you change the current logical oscillator's attack time. To invoke the Select Attack Time option, choose option 12 from the Instrument Definer Menu. The prompt

ATTACK TIME (EVEN NO.) 2 TO 124
=>

will appear on the prompt line. At this point you can abort the command by pressing the ESC key. To complete the command, type an even integer between 2 and 124. When the value has been accepted, it will appear on the Instrument Definer Menu screen to the right of the Select Attack Time option.

Use discretion when assigning attack times. If the music you are playing has notes of very short duration, very long attack times will not work. The sound of the instrument will change drastically if this happens.

Select Decay Time

This option lets you choose the decay time for each oscillator. To choose the Select Decay Time option, choose option 13 from the Instrument Definer Menu. Select Decay Time works much like the Select Attack Time option. You are prompted to type the decay time as an even integer between 2 and 124.

DECAY TIME (EVEN NO.) 2 TO 124
=>

You can cancel the command by pressing the ESC key at this point.

When the Decay Time value has been accepted, it will appear on the Instrument Definition Menu to the right of the Select Decay Time option. The audio feedback will immediately reflect the effect of changing in decay time.

Quit

The Quit option takes you from the Instrument Definition program to the System 2 Main Menu. The System 2 diskette must be in the boot drive or the Quit command will not work properly. If any other diskette is in the drive, or if no diskette is in the drive, you will produce a fatal error. The only way to recover from this error is to reboot. To reboot the MusicSystem, put either of the System diskettes in the boot drive, and press the space bar.

NOTE: Be certain to save your instrument definition before you Quit the Instrument Definer. Once you have left the Instrument Definer, any instrument definition in memory is lost!

Since the consequence of an accidental Quit can be dire, you will be asked to verify your decision.

VERIFY (Y/N) →

Responding with N will abort the command, leaving the current instrument definition intact. Answering with a Y will immediately reboot the diskette in the boot drive.

Special Characters

There are several special characters that are used in the Instrument Definer program. These characters behave differently than you might expect.

The RETURN key behaves a little differently than usual in the coordinate editor. In this special case, it is the signal to incorporate sets of coordinates.

In general, the ESC key returns the Instrument Definer Menu to the screen. The only exception to this rule is when you have chosen option 6, Create Waveform for Oscillator. This option invokes the Wavemaker program in which the ESC key is ignored.

In addition, there are several control characters that have special functions.

CTRL-D is used in the coordinate editor to delete a set of coordinates.

CTRL-H (or the left-pointing arrow, ←) is used in the line editor to delete one character at a time.

CTRL-I is used in the coordinate editor to insert a set of coordinates.

CTRL-M behaves just like the RETURN key.

CTRL-S is used in the coordinate editor to move the cursor down. This character is also used with the Display Statistics feature to scroll the information down.

CTRL-W is used in the coordinate editor to move the cursor up. This character is also used with the Display Statistics feature to scroll the information up.

CTRL-X is used in the line editor to erase the input line.

Chapter IX
MusicSystem Theory

Introduction

This chapter of the MusicSystem manual is the most technically detailed. In this chapter, the general theory behind music and music synthesis is covered. The treatment of this subject alone could easily fill an entire book; the treatment here cannot be as detailed, but the interested reader may refer to additional books on the subject.

After the theory of music synthesis has been presented, the additive synthesis model of musical synthesis is summarized. It is this model that is used in the Mountain Computer MusicSystem. The next topic in this chapter is the Hardware Model. It describes the MusicSystem hardware, and how it operates. The hardware was designed so that it could implement the additive synthesis model in an easy and flexible manner.

Finally, this chapter covers the Instrument Definition Model. This is the software model which drives the hardware to produce sounds. This section, as well as the **whole manual**, is required reading for the user who wants the most performance from the MusicSystem. Concepts essential to the Instrument Definition are introduced and defined. Then the environment in which the MusicSystem works is described. Throughout this chapter, many of the assumptions we have made are noted. These assumptions shaped how we implemented the hardware and software; the MusicSystem is just one approach to the synthesis of sound.

There are many details in this chapter, and it will be worth your time to read it carefully. The information in the Musical Synthesis and the Instrument Definition parts is essential if you wish to create your own instrument definitions, as it represents the theory behind the whole MusicSystem. If you plan to write your own programs for the MusicSystem, you will find the information in the Hardware Model section most helpful. An understanding of musical synthesis is important because our hardware assumes a certain approach to synthesis. It may be helpful to read this chapter once, think about the material, and then read it again.

Definitions

Let's start by defining some terms that we will use throughout this chapter. The definitions will be brief at this point, and they will be expanded at the appropriate point in the text.

What is Music?

Since the object of the MusicSystem is to synthesize music, the first thing to define is music. Exactly what is music? Music is sound, but not just any sound. A sound is usually said to be musical if we can identify a **pitch,** a **loudness** and a **timbre.** Pitch is described as "height" or "elevation". Loudness is described as intensity". Timbre is the "quality" or "color" of a tone. Timbre is the quality of a tone which allows us to distinguish among several different instruments which are all played at the same loudness and pitch. If you were to listen to a piano, a violin, a tuba and an accordion all playing the same pitch at the same loudness, the quality which allows you to identify each instrument is called timbre. A sound is said to be musical if you can ascribe all three of these qualities to the sound. For example, noise has the quality of loudness, but it would be difficult to discern pitch or timbre in noise.

Each of these three qualities of musical sound are identifiable by all people of all cultures. They are each identifiable as a result of the human ear and brain working together. The brain and the ear perform a considerable amount of processing on the sounds we hear. However, each of the three qualities of musical sound have measurable factors which may be perceived by instrumentation. "Pitch" corresponds to the fundamental **frequency,** the number of oscillatons per second of the sound. Loudness corresponds to volume, the **amplitude** of the sound wave. That is measured as the amount of energy in the sound wave. Timbre corresponds to the **spectrum** of the sound. The spectrum is the proportion with which the other, higher frequencies called the harmonics or partials are mixed with each other and the fundamental frequency.

This is a very simple picture; pitch, loudness, and timbre are more complicated than this. If you were to listen to a **pure tone** (i.e., fundamental frequency only), you would perceive a certain pitch. But if you were to change the loudness of the tone, you would perceive a change in the pitch even though the frequency was the same. If you were to listen to a tone of constant amplitude, the perceived loudness would change as you changed the pitch of the

tone. If you listen to a series of short tones at the same amplitude, the duration of the tone will affect the loudness. Duration is the length of time the note sounds.

Timbre

Timbre is even more complicated, because there is more information than just the spectrum which contributes to the perception of timbre. The attack and the decay of an instrument contributes considerably to the timbre. (Attack and decay are parts of the envelope. The envelope describes the amplitude of a single note. The attack is that part of the envelope which describes how, at the beginning of the note, the amplitude starts from zero and increases. The decay is that part of the envelope which describes how the amplitude reaches zero at the end of the note.) This can be demonstrated by playing a tape recording of an instrument backward. The spectrum will still be the same, but you are not able to recognize the instrument because the attack and the decay are reversed. Research has shown that the spectrum of an instrument changes as the instrument is played through its range of pitches. For example, on a reed instrument, playing style will change the spectrum of the instrument. Musical tones are by no means simple, yet they can be adequately described by the pitch, the loudness, and the timbre.

The three qualities described above can be attributed to all musical tones. The question of **what is music** remains to be answered. The attribute which makes musical tones into music is time. If you are in a room and a steady tone begins to sound, at first you will notice it, and then it will become irritating, finally, you will no longer be conscious of the tone, even though it is still sounding. When the tone is turned off, you will suddenly be aware of its absence. In this example, even though the tone was musical (it had a pitch, a loudness and a timbre), the sound was not music. Music is made up of sounds which are physically changing in time. The "physical" changes are in pitch, loudness, timbre or duration. It is important to note that beyond the elements of pitch, loudness and timbre, musical components are cultural. For example, the musical scale of notes; one need only listen to Eastern, Indian and Western music to hear examples of the cultural influence on music.

Rhythm

Rhythm is an important part of music. It is the result of higher functons in the brain, and therefore is influenced by culture. If two pure tones are sounded simultaneously, the brain can single out the tones. It is the superimposing of tones which leads to the concepts of harmony, consonance and dissonance. Again, this is in the realm of cultural influence.

Frequency

Frequency is the number of repetitions of a periodic process in a unit of time. Frequency is expressed in a unit called **hertz** which is one periodic process or cycle per second. The sounds that you can hear will range in frequency from 30 hertz to 18,000 hertz (also 18 kHz). Some people will hear even higher frequencies. When you hit a tuning fork, it will sound at a specific frequency, say 440 Hz. This is because the ends of the tines are moving from one extreme to the other 440 times a second. This causes the air around the tines to vibrate at the same rate. The vibrations in the air cause small changes in the air pressure, which are called sound waves. The rate at which the air pressure changes is directly proportional to the frequency of the sound. The MusicSystem hardware has a different frequency control for each of its sixteen digital oscillators.

Harmonics

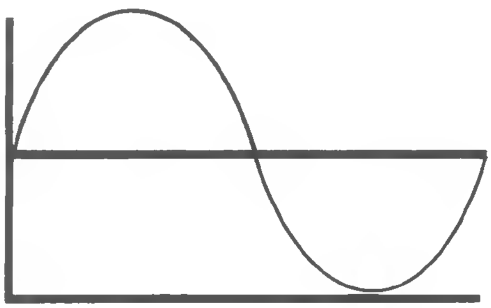
Directly related to frequency is the concept of a **harmonic.** A **harmonic** is an oscillation whose frequency is an integral multiple of the fundamental frequency. What this means is that given **any** frequency, multiplying that fundamenal frequency (in hertz) by 2 (n) will result in the second (n_{th}) harmonic. For example, the third harmonic of the fundamenal frequency is the fundamental frequency multiplied by three. For a frequency to be called a harmonic, the fundamental may only be multiplied by integers.

Practically all instruments, when played at one pitch, will produce several overtones or harmonics. The pitch that you perceive is usually the fundamental frequency. The relative loudness and richness of these harmonics make up the spectrum and hence the timbre. Without these overtones, you could not identify one instrument from another. Careful measurements reveal that overtones are not

related to the fundamental freuency in an integral way. The overtones change their frequency slightly as the instrument sounds one note. These changing "harmonics" are called **partials.** We shall use the terms harmonic, overtone and partial throughout this chapter. Although they will be used interchangeably at some points in the text, be aware that they do not mean exactly the same things.

"Pure Tones"

The "pure tone" was mentioned in the above paragraphs, yet it has not been defined. The **pure tone** that we are talking about is a sinusoidal wave of a fundamental frequency only. It contains no harmonics. The sinusoidal wave is often called the **sine wave.** Below is a graph of one sine wave.



This is a graphic representation of one period of a sine wave. It is also called the **waveform.** If you think of the horizontal axis as distance, and the vertical axis as absolute air pressure, with the center horizontal line at ambient pressure, then the graph shows the instantaneous air pressure at any distance along the horizontal axis. Another visualization is that the horizontal axis is time moving from left to right, and the vertical axis is the displacement of the tuning fork's tine from the center or rest position. Finally, the graph could be the same information that an electronic oscilloscope shows. Time is along the horizontal axis, going from left to right and the vertical axis is the measured voltage (perhaps the output of an oscillator?). The center line is zero volts, above it is positive voltage, and below the line is negative voltage. The waveform is a powerful concept, and will be used in the contexts presented above, as well as other contexts.

Period

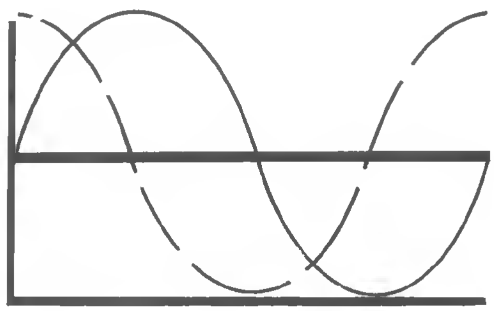
The **period** of a waveform is the time it takes for one full cycle of the periodic process or oscillation. The period is inversely proportional to the frequency of a waveform, i.e.,

PERIOD = 1 / FREQUENCY

Therefore, if the frequency is 2 hertz, the period is 1 millisecond.

Phase

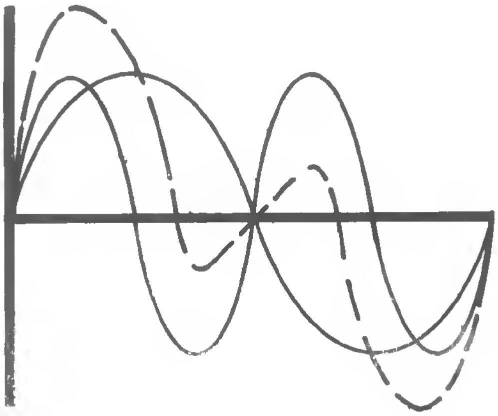
Suppose that you have two sine waves of the same frequency. If the sine waves both begin a cycle at the same time, then the two sine waves are said to be in phase. If the cycles begin at different times, the two waves are said to be out of phase. A **phase** is a particular state in a regularly recurring cycle or oscillation. Phase is measured in degrees, the same way that a circle or compass heading is measured. Below is a picture of two identical sine waves which are 90 degrees out of phase.



When two or more waves are present at the same time, the waves combine to make a resultant wave. This resultant wave is the sum of the amplitudes of all the component waves at one point. Two waves of the same frequency and phase will sum to produce a wave of greater amplitude. Two identical waves which are 180 degrees out of phase will cancel each other out, and the result is zero amplitude. When the frequencies and the phases of the waves being summed are different, the resultant waveform is complex.

In theory, **any periodic waveform may be synthesized by summing only a sine wave and its harmonics.** The relative amplitude of the harmonics can be controlled, as well as the phase difference of all the harmonics relative to the fundamental. Below is a figure of a fundamental sine wave and its second

harmonic. The sum of the two waves is indicated by the heavy line. The concept of summing waveforms is very important to the theory of music synthesis, so if you are not sure how this is done, please reread this section carefully.



Although the summation of sine waves was shown above, you are not limited to sine waves alone. Any two or more waveforms may be summed together to make a new waveform. The MusicSystem is capable of making waveforms of any shape. More details about the programming of waveforms are given later.

Electronic Representation of Music

As you are aware, electronics play a large role in music today. The whole phonographic industry is based on the fact that many people own a stereo system which will play the records they sell. We tend to take for granted the role that electronics play in music today. Music can be represented and manipulated by electronic means. For example, let's examine a tape recorder. A microphone is connected to the tape recorder. There is also a speaker, and the tape. The microphone senses the changes in air pressure which make up sound, and changes those sound waves into electrical waves (voltages) in a wire. These voltages are amplified and intensified by the electronics in the tape recorder, and then stored on tape. When the tape is played back, a reproduction of the electrical waves are amplified and fed to the speaker. The speaker converts the voltages into changes of air pressure or sound. It should be obvious that these are electrical signals which really represent "the music". The MusicSystem can synthesize its own electrical signals which when fed to your stereo and played over the spakers will make music (or many strange sounds if you wish).

ADCs and DACs

Just as there is a way to represent sounds with electronic signals, there are ways to represent electronic signals with numbers in a digital computer. The concept of representing voltages with computer data has applications in many areas. Using numbers to represent the signals which represent sound is the central idea here. A series of numbers in the computer represent a sound. However, there must be a way to convert from numbers to voltages and from voltages to numbers. There are special circuits available for computers called **Analog to Digital Converters (ADC)** and **Digital to Analog Converters (DAC).** If you feed a number to a DAC, a voltage proportional to that number will appear at its output. If a voltage is fed to the input of an ADC, the computer can in turn read a number from the ADC which is proportional to the voltage at the ADC's input.

What if you placed a microphone in a room and hooked the microphone into an amplifier, and fed that to an ADC and a computer? The computer could read numbers from the ADC and place them into its memory for later use. Suppose also that the computer could output numbers from its memory to a DAC, and the DAC voltage was amplified and fed to speakers? If the computer output the same numbers it read from the ADC, you would hear over the speakers what the microphone heard in that room. You would have a very expensive but versatile tape recorder!

ADCs and DACs have a certain resolution specification measured in bits. An eight bit ADC would be able to resolve 256 (2^8) different voltages in its input range, and return a unique number to the computer for each voltage. An eight bit DAC would take any one of 256 possible combinations of eight bits and output a unique voltage proportional to the eight bit number. The MusicSystem uses eight bit DACs, and the numbers fed to the DACs come from the individual entries in the waveform tables.

Filters

The final topic in electronics is filters. A filter is a circuit which will discard any signal meeting certain frequency requirements. A low pass filter will pass only frequencies lower

than a certain frequency. Any frequencies higher than that certain frequency are not passed to the output of the filter. Filters are used to eliminate unwanted signals. The MusicSystem uses a special type of low pass filter to produce exceptional sound quality. The filter aids in the synthesis by smoothing the output of the digital oscillators.

Synthesis and Analysis

It can be shown mathematically that **any periodic oscillation,** however complicated, **can be described as the sum of simple harmonic vibrations.** This is the basis of musical **additive synthesis.** A periodic vibration or oscillation is a pattern which repeats the same motions in the same amount of time. Simple harmonic vibrations are simply sine waves which are harmonically related. Theory states that **any** periodic oscillation can be described as the sum of simple sine waves which are harmonically related.

In musical synthesis, it is desirable to synthesize any musical instrument by producing the same waveform that the instrument does. Most instruments produce periodic oscillations, but a few do not. Drums are an example of an instrument which is not periodic in nature. To produce the same waveform as the instrument produces seems simpler than it really is. First of all, you have to know what the waveform of the instrument looks like. You could wire a microphone into an amplifier and feed the output to an oscilloscope. If you played an instrument into the microphone, the oscilloscope will display the waveform of the instrument. An alternative is to obtain the waveform information from literature. A source of that information will be mentioned, as it will be useful for you when you define your own instruments. Once you know what waveform you wish to synthesize, you are on your way to entering that information into the computer so that the MusicSystem may make the sound for you.

Additive Synthesis

This is so important it will be stated again: Any periodic waveform may be described as a sum of simpler harmonically related waves. The waveform to be described has a definite period. The frequency is the inverse of the period. From the period of the waveform, you calculate the fundamental frequency. This fundamental frequency is summed with harmonics of different amplitudes and phases to produce an approximation of the given waveform. The amplitude and phase factors can differ for each harmonic. These factors contribute greatly to the shape of the resultant waveform. A technique called **analysis** is used to break up a periodic waveform into its component harmonics, their amplitudes and phases. The beauty of the mathematics is that you may go either way; from waveform to components or from components to waveform.

What happens when two waves are summed? Remember the graph of a sine wave which was presented earlier. Think of the horizontal axis as time and the vertical axis as the amplitude of the waveform. The summing of the waves is accomplished this way: for the two waves which are to be summed, simply add the amplitudes at the same point in time (along the horizontal axis) and place the resultant amplitude at the same point in time. Repeat this process for all points along the time axis. When more than two wave are being summed, simply sum the amplitude from each wave to produce the result. It this process is carried out by electronics, the signals to be summed are fed into a summing circuit which wil produce the result on its output. If the waveforms are represented as numbers in a computer, then the computer sums the amplitudes for all waveforms at a given point in time to produce the result.

You could produce a waveform with the MusicSystem by placing a fundamental sine wave in one digital oscillator, its second harmonic in another oscillator, the third harmonic in the third oscillator, and so on. The amplitude control on each oscillator could be adjusted to the correct relative amplitude of the harmonic. There is no phase control on the oscillators. Changing the phase of a harmonic will change the shape of the resultant waveform, but it will not make any significant change on the sound that you hear. So for the purposes of additive synthesis, the phase of all the harmonics can be set to zero. (This is not to say that phase makes absolutely no difference on the sounds that you hear. If you need to generate phase differences between two waveforms, store into a waveform table one waveform starting at a different phase.)

Run Time Calculations vs. Waveform Tables

Once you have set up the oscillators with the fundamental frequency and the harmonics, and have adjusted the amplitude for each oscillator, you could let the electronics of the MusicSystem do the summation of the components to produce the resultant waveform, which drives the speakers. This approach is not efficient because it uses many oscillators to produce one resultant waveform. Since the waveforms for each of the MusicSystem's sixteen digital oscillators are programmable, and we have shown that the summation can be carried out by the electronics or the computer, why not have the computer combine the fundamental frequency with the harmonics at the correct amplitudes. The computer could do the summation once and place the result into a waveform table which would drive one digital oscillator. That one digital oscillator would produce the waveform which resulted from the summation of the components.

You see that the summation which makes up additive synthesis can be done in the electronics with a waste in the use of oscillators, or it may be done by the computer and stored as one waveform to drive one oscillator. However since there are only 256 bytes per waveform table there could be some loss of information in the waveform. This loss can be remedied by using more than one oscillator. If in doubt, you should experiment with this trade-off to obtain the best sound. The MusicSystem has tremendous flexibility because the waveform for each of the oscillators is programmable. This may be taken one level farther because the complex waveforms in the waveform tables are summed by the MusicSystem electronics to produce even more complicated waveforms. For certain sounds, it might be better for you to place separate waves and harmonics in the oscillators. You could have much more control over the individual components of the waveform which would be helpful in generating complex sounds. For other sounds, you will save space by placing the resultant waveforms into waveform tables of their own. As you gain more experience with the MusicSystem, you will know which method is best for a particular application.

The MusicSystem includes a program which will take as input the specifications for the fundamental frequency and the relative amplitude of the harmonics and sum those waves into one resultant wave. You will be able to hear the waveform immediately, and thus you may fine tune the waveforms and harmonics to produce the sound you want. This program is called the INSTRUMENT DEFINER program.

Analysis

The inverse of the summation process is called **analysis.** In analysis, you are given one period or cycle of a periodic waveform. Your task is to determine which components (harmonics), and what amplitudes of the components will produce the input waveform when summed together. This is the reverse process of additive synthesis. There is a lot of mathematics for this analysis, and it is no surprise that the computer is used as a tool in the analysis of sound. Think of it as a black box which contains the mathematics for analysis. Into the black box you feed a numerical representation of sound. Then you obtain from the black box the harmonic components and the amplitudes which will sum to synthesize the original sound. These same components could be fed into the inverse process to produce the complex waveform (additive synthesis).

There is a lot of research being done on the analysis of musical sounds. This research has been helpful to us in the development of the MusicSystem, and it can be helpful to you when you define your own instruments. A reference to this literature is cited in the Bibilography. It is useful to know how the analysis of the sounds was accomplished. But first, a few things must be explained.

Sampling Theory

REcall the discussion about ADCs and DACs. In an earlier example, it was shown how a microphone could feed an amplifier which feeds an ADC which produces numbers for the computer. The numbers produced by the ADC are directly related to the voltage or amplitude of the electrical signals fed to the ADC. These numbers in the computer constitute a numerical representation of the sound picked up by the microphone. Since they are numbers in a computer, they may be processed in many different ways.

But how many numbers do you need to represent a sound? And how big must those numbers be? How many numbers you need

depends on the highest frequency of sound you want to represent. As you increase the frequency, the quantity of numbers per unit of time increases also. The size of the numbers is determined by how accurately you would like to specify the amplitude of the sound.

Remember that the number of bits that the ADC outputs to the computer defines its resolution. Eight bits of resolution at the ADC means that the ADC can resolve 256 different voltages at its input. The voltages must be in a certain range of voltages (for example, between minus one volt and plus one volt) which depends on the particular ADC circuit. Within that range, the eight bit ADC can resolve 256 different voltages. A twelve bit ADC can resolve 4096 different voltages in its input range. Regardless of the resolution of an ADC, the ADc will output the number which represents the voltage **closest** to the input voltage. There is a small amount of error, called the **Quantization error,** which is equal to the smallest step of voltage the ADC will resolve. The higher the resolution of the ADC, the smaller the voltage step and the smaller the quantization error.

This is how a microphone fed to an ADC can be used to input sounds to the computer. The computer reads the numbers output by the ADC, and except for quantization error, the number represents the input voltage. In this way, the computer can read the instantaneous amplitude of the waveform presented to the microphone. We mentioned that the highest frequency also determines the quantity of samples needed per unit time to represent the sound. At this point we enter the realm of sampling theory. When the computer reads the numbers from the ADC, it is taking samples of the waveform at that instant in time. The number of samples that it reads in a unit of time is called the **sampling rate.** This is expressed as samples per second, for example 100 samples per second. Of course, it takes some small but finite amount of time for the computer to read the number from the ADC. Indeed, the ADc will require a finite length of time to convert a voltage into a number. The computer usually commands the ADC to start a conversion, and then a short time later, the ADC will return a number to the computer. Therefore, there is a maximum number of samples which may be read from the ADC in one second of time. The maximum number depends on how fast the ADC is and how fast the computer can store a number and get ready to read the next number. If a fast ADC and computer can convert a voltage to a number and store that number somewhere in memory in 100 microseconds, then the maximum number of samples which may be read in one second is the reciprocal, that is, 10,000 samples per second.

Sampling theory states that to accurately represent a waveform whose highest frequency is (N) Hz, you must take **at least** 2 times N samples per second. In other words, **the sampling rate must be at least twice the highest frequency you wish to sample.** If the highest frequency you were interested in was 16kHz (16,000 Hz), then your computer must sample the input voltage (the waveform) at least 32,000 times per second. Severe distortion of the waveform and loss of information results when the sampling rate is not high enough. Digital recording techniques used in the record industry have a sampling rate around 50,000 samples per second. This means that the ADC and the computer together must process one sample every 20 microseconds. They can accurately reproduce a frequency up to 25 kHz. Another aspect about sampling theory is that the samples must be taken periodically. This means that the time between samples must be precisely 20 microseconds. Samples taken at nonperiodic times will not represent the waveform at all. For sampling theory to work at all, the samples must occur periodically.

In the MusicSystem hardware, each digital oscillator gets a new sample at the rate of 31,250 samples per second per oscillator. This means that each oscillator may synthesize a frequency up to 15,625 hertz. The term **Nyquist frequency** is defined as one half the sampling rate. Thus the Nyquist frequency for the MusicSystem is 15.625 kHz.

Foldover

If you try to produce a frequency on a digital oscillator which is greater than one half of the sampling rate, distortion occurs. This distortion shows up as a frequency distortion. There are unwanted frequencies in the output of the synthesizer when foldover occurs. A foldover is sometimes called an **alias.** Since the output of the oscillator goes through a low-pass filter on the MusicSystem board, the high fre-

quency you were trying to generate will not get past the filter. But the alias will get through, and introduce frequencies into the output which were not specified.

For example, suppose that the waveform table of one of the digital oscillators is loaded with a simple sine wave. The frequency register is then set at a low frequency. Suppose that throughout this experiment, you could hear the sound through speakers. Now as you start to increase the number in the frequency register, the pitch of the pure tone increases, and everything sounds fine. As the frequency of the oscillator approaches the sampling frequency, everything goes well. Now suppose that the frequency register now specifies a frequency a little bit higher than the sampling frequency. Instead of hearing a steadily increasing pitch, the pitch will go up to a point, and then start to descend. The sampling frequency acts as "mirror" to frequencies greater than its value. If you wish to generate the waveform at frequency F_g (the frequency of generation), and the sampling frequency is called F_s, then the frequency of the resultant output, F_r, is specified by this formula:

$$F_r = F_s - F_g$$

For example, if you tried to generate a frequency of 20 kHz, and the sampling frequency was 32 kHz, the resultant frequency would be 12 kHz. If you were trying to generate a frequency of 30 kHz, the resultant frequency would be 2 kHz. This "mirror" action accounts for the name "foldover."

If you create a complex waveform which contains many harmonics of the fundamental frequency, the **higher** harmonics may foldover and produce unwanted frequencies. Suppose that the waveform had a fundamental frequency of 10 kHz, and only the second harmonic at 20 kHz. The fundamental frequency would come out correctly, because it is less than the Nyquist frequency. But the second harmonic would be heard at 12 kHz, which is very near the fundamental frequency. This is not what was desired! If there were a third harmonic in the waveform (30 kHz), it would appear at 2 kHz, and the fourth harmonic (40 kHz) would appear at 8 kHz. Now imagine that the fundamental frequency is lower and there are 12 harmonics in the waveform. The results of playing that waveform would be very strange indeed! This is great if you are making a Martian xylophone, but in general, foldover should be avoided. Pay attention to the frequencies you generate and the harmonic content of your waveforms.

Analysis of musical sounds can be done with the aid of computers. Usually, a microphone is wired to an ADC which is wired to a computer. The sound to be analyzed is played into the microphone while the computer collects data. The sound played is most often a single, steady note. This note should have a steady frequency and amplitude and should last for a duration of about one second. This analysis technique will work for instruments which produce a periodic waveform only. The computer will analyze the sound and print out (or perhaps draw out) the results of the analysis.

Consider that the source of data for the analysis is one single pitch, one note. This analysis should yield the harmonic content of the waveform, based on a fundamental frequency, and it would be nice if it gave you some int rmation about the envelope of the note. You recall that the envelope and the spectrum together play an important part of defining the timbre. The **envelope** is the amplitude of the note as the note builds up (**attack**), as it holds (**sustain**) and as it dies down (**decay**). The envelope is treated in more detail in the Instument Definition section of this chapter.

The envelope specifies how the amplitude of the note behaves. Also important are the frequency changes during the attack and the decay. This is the notion of a partial, or a harmonic which is not exactly an integral multiple of the fundamental frequency. As you will see in the Instrument Definition section, the frequency control on the MusicSystem hardware specifies both the pitch of a note and the frequency as it changes during the note. We call this the "frequency history" of a note. The envelope is called the "amplitude history."

Since the analysis is based on the playing of a single note, it seems reasonable that the synthesis should produce a single note. Consider the "note" as the basic object of the synthesis process. This is a limitation which makes the Instrument Definition conceptually easier to understand. It is **not** a limitation on the power of the MusicSystem.

Harmonics and Timbre

Are the harmonics alone responsible for the timbre? Is spectrum all you need to specify a timbre? The answer "No." This information is repeated here because it is so important. The spectrum and the amplitude envelope are very important for defining a timbre. You should try using the INSTRUMENT DEFINER program and changing the amplitude history of an instrument. Experiment with the envelope control and hear how much difference the envelope makes on the timbre. Likewise, experiment with the frequency history and the harmonics. All of these elements are important for defining a timbre. There is much that is not known about how humans recognize and perceive different timbres.

Pure additive synthesis allows for a phase specification for each harmonic. The shape of the resulting waveform will vary as the phase of even one of the harmonics varies. However, most of this phase meaning is lost on the human ear. Therefore, the MusicSystem has no direct provisions for controlling the phase of the oscillators. If all oscillators were at the same frequency, and generating the same waveform, then all the waveforms would be in phase. If you need to introduce a phase difference, load one oscillator with a reference waveform, and then change the phase by some degree and store that new waveform into a second oscillator. The two oscillators played at the same frequency will have a phase difference equal to the phase difference of the two waveform tables. Remember that for the most part, phase control will not make a detectable difference in what you hear.

Summary

Several basic concepts have been discussed in this section. You should be sure that you understand these ideas before reading farther. In this section we discussed synthesis and analysis. The concept of additive synthesis was introduced. Sampling theory was touched upon, as well as the technique of analysis. The data from analysis is for a single note, played at one pitch. This spectrum data along with the amplitude history and the frequency history defined the timbre for one note. Usually this data is applied to a range of notes. However this may not be appropriate, as the spectral content, or the amplitude history, or the frequency history or any combination of the three will be different when the instrument is played at a different pitch. And of course, the synthesized instrument can be played in a pitch range very different from the original instrument, although it may not always sound like the original instrument.

There is much research being done in this field today. There are many unanswered questions about music and sounds. The MusicSystem offers a great device for exploring this region of communications. Perhaps a clever person will use the Mountain Computer A/D + D/A board along with some software to capture and edit waveforms from a microphone or other input. The ultimate would be to do the analysis with the Apple and use the results to drive the MusicSystem.

The Hardware

Introduction

The Mountain Computer MusicSystem consists of two boards which plug into two consecutive slots of the Apple II computer. The two boards are joined by a ribbon cable. The MusicSystem develops a stereo output via a dual standard RCA type phono connector, the type on most home stereo systems. The MusicSystem includes a light pen which is used in the Music Editor. The light pen may also be used for other applications.

The MusicSystem hardware implements sixteen independently programmable digital oscillators. The operation of the oscillators in making music is covered in "The Hardware Model." These digital oscillators are said to be programmable because the waveform output from the oscillator is user specified. The waveform may be arbitrarily simple or complex.

The MusicSystem uses a waveform table which is 256 bytes long. Therefore, any periodic functon of time (waveform) that may be expressed as a set of 256 values may be programmed into any one or all of the sixteen digital oscillators. The sixteen digital oscillators are grouped into two groups of eight. The outputs of the eight oscillators in a group are summed together and then filtered and buffered before going to one of the phono connectors. One group of eight oscillators summed together produce the LEFT channel

output, and the other group produces the RIGHT channel output. Together they make the stereo output of the MusicSystem.

There are additional controls implemented by the hardware. An overall volume control for the output from the MusicSystem which affects all sixteen oscillators, a control for enabling or disabling interrupts and a control for enabling or disabling the MusicSystem boards. There is also a control that lets you check the status of the light pen and read a random 5-bit number.

In the Hardware section the waveform table, its use, its format and interpretation are discussed. There is a general theory of operation for the programmable digital oscillator and the board in general. The mechanics of programming the oscillators and controlling the board are discussed. There is a reference section and a memory map of the control register addresses.

The Hardware Model

This section contains details about the hardware of the MusicSystem. The information is called "The Hardware Model" to remind you that this hardware represents but one of many ways to make an electronic device which is capable of synthesizing complex sounds and music. The process of synthesizing music is discussed in the Musical Synthesis section. If you have not done so, read that section. This discussion about the Hardware Model assumes that you are familiar with the terminology used there and the mechanics of the additive synthesis model for musical synthesis. The hardware has been built in such a way as to fit nicely with that model of synthesis.

An overview of the Hardware Model shows the sixteen programmable digital oscillators (in two groups of eight), the filter, the buffer, the global controls and the output. The overview of the hardware is diagrammed here.

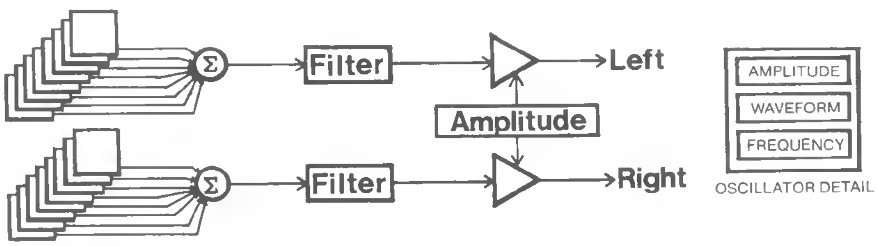
In the figure below, the left and right channels of the stereo output are produced by summing the outputs of eight oscillators, filtering that resultant sum, and amplifying the resulting waveform for delivery to your stereo inputs. Each oscillator has a separate amplitude, frequency and waveform table address control as indicated by the detail of one oscillator.

Digital Oscillators Summed to Output a Waveform

The outputs from eight oscillators are summed together producing a complex waveform. The components of the waveform are the outputs of the oscillators. The summation performed by the electronics is similar to the additive synthesis technique. Particular oscillators are assigned to one of the two groups. This assignment places the even numbered oscillators on the LEFT stereo channel and the odd numbered oscillators on the RIGHT stereo channel. This assignment is fixed and cannot be changed.

Filtering

After the outputs are summed, the synthesized waveform is fed to a filter. The filter must process the waveform before it is passed through the filter. The filter on the MusicSystem is designed to prevent the output of certain frequencies to the phono jacks. The design considerations involve the sampling rate and the elimination of "aliasing." These topics were discussed in the section on Musical Synthesis. Normally the filter will not be of concern as you will adjust the synthesizer for the sound you want. Be aware that if you are doing unconventional things, for example, placing a lot of amplitude on the 20th harmonic of a waveform and then playing the waveform at a high frequency, you will produce some high frequencies, and the filter will definitely affect the output. Let your ear be the judge.



Buffering the Output

The output from the filter is fed to the buffer amplifier. The buffer does two things to the waveform. First it serves to boost the signal's power to a level suitable for the inputs of a stereo, and at the same time it isolates the synthsizer from the stereo. Secondly, the buffer is controlled by the overall amplitude control. This control is a one byte (8 bits) digital hardware register. Into this memory location, a number in the range of 0 to 255 is written, directly controlling the overall volume of the stereo outputs. 0 means the lowest volume, as if the volume on the stereo were turned all the way down. 255 is the highest volume, as if the stereo were turned all the way up. The output of the buffers is fed to the phono plugs.

Independent Controls

Each of the 16 digital oscillators has its own set of independent controls. These controls are implemented in hardware as WRITE only memory locations in the MusicSystem boards' I/O space. Within this chapter, a table presents the addresses of the control registers for all oscillators. The registers for each oscillator are the amplitude register, the frequency register, and the waveform table address register. Because these controls are independent for each oscillator, it is possible to have 16 different waveforms, each at a different volume and pitch, playing simultaneously.

Amplitude Control

The amplitude register works just like the overall volume control except that it controls the volume of its particular oscillator only. This register is usually used for envelope control on a waveform, although it may be used for any purpose. Just like the overall volume control, 0 is the lowest volume and 255 is the highest volume. The register is a write-only location so the current value of the amplitude must be elsewhere in memory if its value must be known.

Frequency Control

The frequency register is two bytes long. The two bytes are called FREQ H and FREQ L for the high and low order parts of the frequency value. The FREQ H part is one byte, an integer from 0 to 255. The FREQ L part is one byte which is treated as a fractional part of the frequency value. Together the integral and fractional parts specify the frequency. Although the value of the integral part of the frequency register may take on values from 0 to 255, a value above 128 does not make much sense. The reason for this will be explained later. The lower part of the frequency register, FREQ L, that there is a decimal point to the extreme left of the FREQ L byte. Then the place values of the digits going towards the right are 1/2, 1/4, 1/8, 1/16, 1/32, 1/64, 1/128, and 1/256. Therefore a fraction like 3/4 (0.75) could be expressed as the binary number (.)11000000. The decimal in parentheses is imaginary. The ability to specify a fractional part of the frequency allows for fine frequency control of the oscillator's waveform. The frequency register is a write-only register.

Waveform Table Addresses

The waveform address register is a one byte write-only register which contains the page address of the waveform table for a particular oscillator. Each waveform table is 256 bytes long and all waveform tables must begin on a page boundary. In brief, the Apple's memory space is 65,536 bytes (also called 64K). This can also be expressed as 2^{16}, or $2^{8 * 8}$. The 6502 microprocessor in the Apple II has 16 address lines. You can think of the address lines as two bytes of address.

The paging idea makes sense when you think of the upper byte of the address as the address of one page of memory. The lower byte then is the address within that page. Since a byte has 8 bits, 2^8 is 256 and each page holds 256 bytes. There are a total of 256 addressable pages in the available memory space. Not all of the pages are RAM memory in the Apple II. Some pages are used for Read Only memory and cannot by used for programmable waveforms. Notice the term "programmable." There is nothing to prevent you from playing the Apple II motherboard ROMs as a waveform. If you want to use your own waveforms, they must be located on a page of memory in RAM. The waveform address register is written with the page address of the waveform table for that oscillator. It is a write only location.

Each of the three control registers for the digital oscillators may be changed as often as once per 32 microseconds when the MusicSystem is enabled and playing music.

Global Controls

The next item in the hardware model is a set of global controls. These controls affect the operation of the MusicSystem boards and, in the case of the volume control, all sixteen oscillators.

There are a total of four global control registers. One register serves as the light pen status bit and a 5 bit random number generator. A second register controls the overall volume or amplitude of the MusicSystem's output. The third and fourth control registers actually map into two memory addresses each. It is useful to think of the last two registers as flip-flops. A flip-flop is a 1 bit hardware register. These registers are used to enable or disable the interrupts and the DMA.

The Light Pen and Random Numbers

The light pen status and random number generator shares the same address in memory as the overall volume register. When a READ operation (such as a PEEK) is performed at the address, the light pen status is returned in the most significant bit (bit 7), and a random 5 bit number appears in the lowest 5 bits (bits 4-0). The light pen bit is a 1 (the PEEK returns a value greater than or equal to 128) if the light pen is "seeing" light. The light pen bit is a 0 (the PEEK returns a value less than 128) if the light pen is not "seeing" light. To obtain the random number, mask off the high order bit. If the PEEK returns a value less than 128, that value can be used as the random number. If the PEEK returns a value greater than or equal to 128, subtract 128 to get the random number. Please note that the random number is updated only when the MusicSystem is enabled (when the DMA is on). DMA is discussed in the section describing the operation of the oscillators.

The random number is produced by the DMA controller. The number is the low order 5 bits of the address of the last sample of the last waveform table. When the MusicSystem is enabled the random number changes once every two microseconds. If the MusicSystem is disabled, the random number will not change.

Overall Amplitude Control

The overall volume control register shares the same memory address as the light pen and random number register. When a WRITE operation is performed to this address, the number written is stored into the overall volume control register. The value written into the register may range from 0 to 255. The value 0 is the lowest volume, and the value 255 is the highest volume. As seen in the hardware model diagram, this register controls the overall level of the right and left channels to the stereo.

Synthesizer Enable/Disable Control

The MusicSystem board enable/disable register is used to turn the MusicSystem ON and OFF. When the MusicSystem is enabled, DMA is taking place. When the board is disabled, the DMA is not operational. This register actually controls the DMA. In order to activate the board, the user must WRITE to the ON address TWICE. The WRITEs must occur within 100 microseconds of each other. To deactivate the MusicSystem boards, the user must WRITE to the OFF address ONCE. A RESET also turns the MusicSystem OFF. You must NOT READ either the ON or OFF addresses. The actual addresses are slot dependent and are specified in the reference section of this chapter. One important implication that occurs when the MusicSystem boards are enabled is that of the DMA. When the DMA is on, it is using every other memory cycle of the Apple II computer. The other memory cycles are used by the 6502 CPU for running programs. Since the DMA uses every other memory cycle, the processor runs programs at half normal speed.

Interrupt Enable/Disable Control

The last global control register is used for enabling and disabling the interrupt generator on the MusicSystem board. There is a jumper area on the board used for the selection of the interrupt rate. It is factory set at the 8 millisecond rate. The MusicSystem software expects this rate of interrupts. You should change this jumper only if your own software requires it! The MusicSystem software uses interrupts for timing purposes. The interrupts are independent of the operation of the DMA. To enable the interrupts you must READ from the ENABLE address. To disable the interrupts you must READ from the DISABLE address. You must not WRITE to either of the interrupt register addresses.

Those of you who write your own software to control the MusicSystem hardware should bear in mind that because of the demands of the DMA no other device should use the DMA

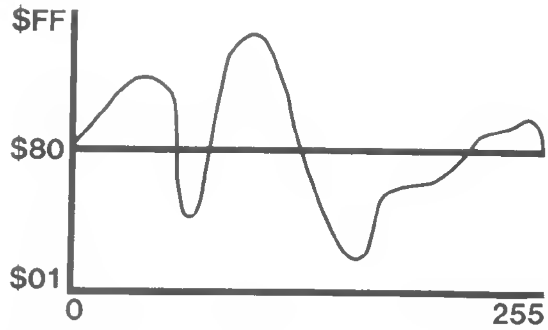
while the MusicSystem hardware is enabled. Likewise, no other device should be generating interrupts while the MusicSystem hardware has its interrupts enabled.

The Waveform Tables

The hardware model presented assumes that the data for the periodic waveform of an oscillator is arranged as a table of 256 bytes. This waveform table must begin on a page boundary in memory. In other words, the beginning address of 256 consecutive bytes which represent the waveform must be an integral multiple of 256. If the addresses of the waveform tables were written in hexidecimal, the last two digits must both be equal to a zero for a waveform table to start on a page boundary. This restriction exists for this reason: if the tables are 256 bytes long and the address must start on a page boundary, the page address will remain the same throughout the address space of the waveform table. Thus the waveform table must be 256 bytes long and begin on a page boundary.

Waveform Table Format

The waveform table usually represents one period of the waveform to be produced by the oscillator. Exactly how is this table interpreted? We have found that 256 numbers or values can describe many different waveforms. The waveform table as a whole makes up one period of the waveform. That one period is closely approximated by 256 numbers. Those numbers represent the amplitude of the waveform at one of the 256 sample times. While an oscillator is running it gets the next entry from the waveform table and converts that number into an amplitude or voltage. Sounds can be represented in electronic circuits by voltages, and voltages can be represented in computers by numbers. If the value in the waveform table entry is $80 hexidecimal, the output voltage is 0. If the value in the entry is $01 hexidecimal, the output voltage is at its maximum negative value. If the value in the table entry is $FF hexidecimal, the output voltage is at its maximum positive value. The following graph shows a waveform table.



Waveform Table Interpretation

The x-axis goes from 0 to 255 and corresponds to the entry number in the table (an address in memory). The y-axis goes from $01 to $FF. At $80 along the y-axis is a horizontal line which represents the output voltage of zero. Above this line are 127 positive values, and below this line there are 127 negative values. The waveform may take on any numeric value from 1 to 255 at any of the 256 different entries. The numeric value of zero is not used to generate a waveform table. If one of the entries should contain a value of 0 the result is maximum negative output, as if the value was 1.

Creating Waveform Tables

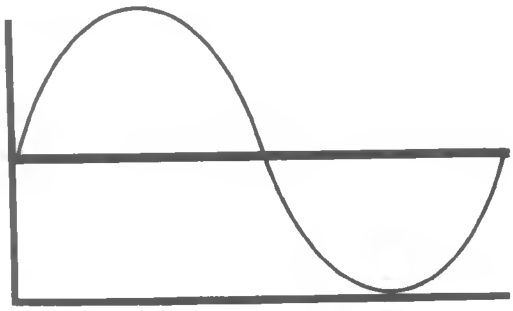
There is nothing special about making a waveform table. You should experiment with making and playing your own waveforms. Later in this chapter, a simple program is included which will produce 256 byte waveform tables and save them to the diskette. Any 256 byte binary file may serve as a waveform table when loaded at a page boundary.

When making your own waveform tables there are a few things to know. Whatever function or waveform you represent in the table, at least one of the values should be either an $FF or a $01. If this condition is not met, the amplitude of the waveform will never reach a maximum. If you combine it with other waveforms or use the waveform with the INSTRUMENT DEFINER program, you might not get the amplitude results you expected.

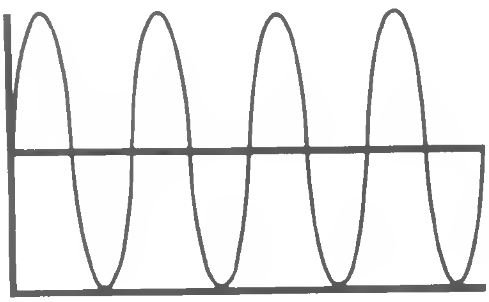
When you create your own waveform table files, you should place the waveform in memory so that it starts on a page boundary. Use the DOS BSAVE to save the waveform table. When saving the waveform table with

the BSAVE command, you MUST specify the file length as L$100 bytes. You must also use the naming convention for waveform files. All waveform files must begin with the characters "WAVE.". A correct example would be the waveform file named "WAVE.RAIN". If this naming convention is not followed, the files cannot be used with the MusicSystem or the INSTRUMENT DEFINER Program.

Since a waveform table is composed of 256 entries, a situation can arise in which the 256 points are not enough to describe a waveform accurately. Take a look at this graph.



There is one period of a sine wave in this waveform table. The 256 points along the x-axis which are each an entry in the table are enough to represent the waveform very well. If you were to place four sine waves into the same waveform table, each sine wave would have to be described by only 64 points. This is how it would look.



If you were to change a waveform table with one sine wave in it to a waveform table with four sine waves in it, the pitch or frequency would be four times greater. The resolution of the waveform is not as great in the latter case, but it usually does not make a difference. The filter in the MusicSystem works to fill in some of the missing information, and the human ear also fills in some of the missing points along the curve. Even though the waveform, as drawn on paper, does not look right, the waveform sounds just fine. Since you want to make sounds, sound is what counts, not looks.

DMA Implications

The waveform tables are in the Apple's memory. Each 32 microseconds every one of the oscillators will need to read a new entry from its waveform table. The DMA takes care of getting the entries for all sixteen oscillators. With sixteen oscillators, it is obvious that the DMA must be fast. When the MusicSystem is enabled it must get one entry every two microseconds. The formation of the DMA address is covered under the operation of the digital oscillator.

How Does It Make Sounds?

You can generate any periodic waveform simply by summing sine waves at the fundamental and harmonic frequencies. A different weighting or scaling factor is applied to each harmonic as it is summed, and this factor represents the amplitude of that part of the sound spectrum. Also remember that the three basic attributes of music are pitch, loudness, and timbre. It is the spectral quality of the waveform and envelope control that are essential for defining the timbre.

The notion of periodic waveforms (for example sine waves), corresponds with the programmable nature of the digital oscillators. The waveform table defines the periodic time function for the oscillator. The waveform could be a simple sine wave, or to further compact data the waveform table could contain the result of the additive synthesis, i.e., several partials already summed into the complex waveform. This allows very complex spectra to be represented in a few waveform tables.

Each digital oscillator has an amplitude control and a frequency control. The envelope of a note can be implemented with the amplitude control. The attack and decay portions of the envelope are critical for producing timbre. The amplitude control is also used to control the loudness of an oscillator.

The frequency control directly affects the pitch at which you perceive the periodic waveform from the digital oscillators. The higher the frequency, the higher the pitch. Notice that there are direct electronic counterparts for pitch, amplitude and spectrum in the digital oscillator.

The Digital Oscillator

What is an Oscillator?

An oscillator has a pattern which repeats with time. Examples of oscillators are flashing traffic lights, a pendulum, a piano string, a guitar string, or even the seasons of the year. Because the patterns repeat with time, the oscillator can be said to describe a periodic function of time.

Oscillators can also be made of electronics. It's easy to build an electronic oscillator whose output voltage describes a sine wave. A digital oscillator is one in which the periodic waveform function may be programmed. The waveform is stored in a table and the table represents one period of the function. Each table has 256 entries.

What is DMA?

When the MusicSystem is enabled, the DMA controller is turned on. DMA stands for Direct Memory Access and is just a mechanism for getting data into and out of memory without having to program every transfer with the microprocessor. DMA is used when a large volume of data must be moved fast. DMA operation is transparent to the processor. That is, it does not interfere with the processing except for introducing some delay. In the MusicSystem the DMA is used to access entries from the waveform tables stored in memory. In normal operation, the parameters for the oscillator and global parameters are initialized. Then the MusicSystem boards are enabled, and the DMA starts.

Sampling Rate and Frequency

Each oscillator gets a new entry from its own waveform table once every 32 microseconds. Since there are sixteen oscillators to service, the DMA must fetch one new entry every 2 microseconds. The sampling rate is the rate at which new samples (entries) are supplied to the MusicSystem. Once every 32 microseconds is the same as 31,250 times a second. Sampling theory states the sampling rate must be at least twice as high as the highest frequency you wish to synthesize. Therefore 15,625 hertz is about as high a frequency as the MusicSystem can faithfully synthesize.

Waveform Table Entry Pointers

The format of the waveform table aids the DMA's job. Since the page address of a waveform table will not change as the oscillator steps through the table, the DMA simply places the page address of the oscillator being updated out onto the upper 8 bits of the address lines. The lower 8 bits of the DMA's address come from a waveform table entry pointer, or "pointer" for short. Each of the sixteen oscillators has its own pointer into its own waveform table. Every 32 microseconds the DMA will fetch the entry from the table to which the pointer points and feed that entry into a DAC. The DAC converts the value at that table entry into a voltage, which produces the oscillator's output.

After the entry has been fetched, the pointer for the oscillator is updated to point to the next entry to be fetched from the waveform table. This entry is dependent upon the contents of the frequency register. The frequency register is composed of a fractional part and an integral part. The new value of the pointer is calculated by adding the frequency register to the current value of the pointer. Internal to the hardware, the pointer to the next entry is maintained as a sixteen bit value. The upper byte is an integer and the lower part of the pointer is a fractional part, just like the frequency register's fractional part. The frequency register's fractional part is added to the pointer's fractional part and the result is placed into the pointer's fractional part. If the sum of the fractions is greater than one, an overflow to the integer part is generated. Next, the integral part of the frequency register and the integral part of the pointer (and the carry, if any) are summed, and the result is placed into the pointer's integral part. This integral part is the next entry in the waveform table. It is a direct index into the waveform table. If this index exceeds the value 255, it simply wraps around to zero.

For example, suppose the frequency register contains a value of 5.0 and the pointer for the oscillator is equal to 254.0. After the next sample is obtained, 5 is added to 254 to produce 259 which is greater than 255. A MOD 256 function is applied and the result of 3 is placed into the pointer.

In another example, the frequency register contains the value 0.25 and the pointer is starting at 0. It will take 4 updates to the oscillator before the pointer changes from 0 to 1. The fractional parts are maintained so that frequencies may be specified accurately.

Frequency Control

The notion of how frequency control works should be made clear. In the last example, the oscillator was updated four times before the pointer advances by one. Since the oscillators are always updated every 32 microseconds, it will take four times longer for this oscillator to step through its entire waveform table than if the frequency register had been set to 1.0. On the other hand, if the frequency register were set to 2.0, every other entry would be obtained from the table. If the frequency were set to 3.75, then the waveform table would be stepped through in an uneven way. Again, sampling theory and the filter help to even out the waveform. In an extreme example suppose the frequency register were set to 128.0. In this example the digital oscillator would only get the entries numbered 0 and 128. This would result in a severe distortion of the waveform in the table. It doesn't make much sense to set the frequency register to a value equal to or higher than 128.

To summarize, when the MusicSystem boards are enabled the DMA is on and each of the digital oscillators gets a new sample from its waveform table every 32 microseconds. The sample which is fetched is converted into the output for that oscillator and that output is scaled by the oscillator's amplitude register. The pointer to the next waveform table entry is updated with the current contents of the pointer added to the value of the frequency register. This process of repeatedly fetching samples continues as long as the MusicSystem is enabled.

Reference Section

In this section, the memory map to the MusicSystem hardware is detailed with all addresses given in hexidecimal. All hexidecimal numbers will be prefixed with a $ symbol. The techniques used to address and read or write the registers will be discussed.

Slot Dependent Memory Addresses

The MusicSystem boards use memory addresses in the I/O address space of the Apple II computer. The MusicSystem may occupy any two consecutive slots in the Apple computer. When speaking of which slot the MusicSystem is in, the lower number slot contains the analog electronics and the cable to the light pen. This slot will be named slot "S". The other board will go into the slot numbered S+1. Slot S cannot equal slot 0 and it cannot equal slot 7. Slot 0 is reserved for firmware cards and slot 7 cannot be used because only half of the MusicSystem would be plugged in.

The Light Pen and Random Numbers

The hardware uses slot dependent addresses in the I/O address space. The first of these register addresses is the light pen status and random number register. This register is accessed by a read to the address $C(S)00, where S is the slot number as described above. The high order bit is a 1 if the light pen sees light, and a 0 if the light pen does not see light. The lower 5 bits make up the random number. The random number changes once every two microseconds when the DMA is enabled. The number does not change at all if the DMA is disabled.

Overall Amplitude

The next register is the overall volume control register. The address of this register is the same as the above address. A write to the address $C(S)00 will program the overall amplitude of the left and right channels. A value of zero is the lowest volume and a value of 255 is the highest volume.

Enable/Disable the MusicSystem

The MusicSystem board is enabled by WRITING to the address $C0(S+9)1 TWICE. The MusicSystem is disabled by WRITING to the address $C0(S+9)0 once. The value written in either case does not matter. It is the WRITE reference to the address that is important. When enabling the MusicSystem, there is a restriction on the TWO writes which must be preformed. The second of the two writes MUST occur within 100 microseconds of the first write. Additionally you may not turn the MusicSystem OFF any sooner than 500 microseconds after the MusicSystem is turned ON. A RESET on the Apple computer will turn the MusicSystem OFF. You must NEVER READ these two locations.

6502 Machine Language Enable/Disable

When using 6502 machine language instructions to enable and disable the MusicSystem, you may execute two STA to an absolute

address instructions in a row to enable the MusicSystem. You may also use one STA to an indexed address to enable the MusicSystem. In the 6502, an indexed instruction does a phantom memory cycle to the address before the write is done. It has the same effect as two references to the address in a row and will enable the MusicSystem. We prefer the use of the STA to an absolute address instruction.

Interrupts

The interrupts which are generated by the MusicSystem may be enabled by READing from the address, \$C0(S+8)1, once. The interrupts may be disabled by READing from the address, \$C0(S+8)0, once.

The following chart presents the addresses of the amplitude, frequency and waveform registers for all sixteen oscillators. All locations are WRITE only and are to be prefixed by the value, \$C(S+1)xx.

ALL ADDRESSES ARE WRITE ONLY!!

ALL ADDRESSES ARE PREFIXED BY \$C(S+1)xx

OSC #	AMP	TABLE	FREQ H	FREQ L
0 (L)	21	20	00	1F
1 (R)	23	22	02	01
2 (L)	25	24	04	03
3 (R)	27	26	06	05
4 (L)	29	28	08	07
5 (R)	2B	2A	0A	09
6 (L)	2D	2C	0C	0B
7 (R)	2F	2E	0E	0D
8 (L)	31	30	10	0F
9 (R)	33	32	12	11
A (L)	35	34	14	13
B (R)	37	36	16	15
C (L)	39	38	18	17
D (R)	3B	3A	1A	19
E (L)	3D	3C	1C	1B
F (R)	3F	3E	1E	1D

OSC #: contains the oscillator's number in hexidecimal and the channel to which it is assigned. "L" is the left channel and "R" is the right channel.

AMP: is the address of the amplitude register for the particular oscillator. The allowed values range from 0 to 255. Zero is the lowest amplitude, and 255 is the highest amplitude. This value is WRITE only and the two digit number in this column should be prefixed by the value \$C(S+1) to obtain the four digit hexidecimal address.

TABLE: is the address which contains the page address of the waveform table for that particular oscillator. This is an 8 bit register so allowed values range from 0 to 255. All waveform tables must be 256 bytes long and must begin on a page boundary. Since this one byte controls the selection of a waveform table, only this byte needs to be changed to specify a different waveform table. Likewise, all oscillators could use the same waveform table. This value is WRITE only and the two digit number in this column should be prefixed by the value \$C(S+1) to obtain the four digit hexidecimal address.

FREQ H: is the address of the high order (the integral) part of the frequency register. The allowable values range from 0 to 255, but, in practice, the upper limit to the value is 128. The frequency register can be thought of as an increment for the pointer into the waveform table. This value is WRITE only and the two digit number in this column should be prefixed by the value \$C(S+1) to obtain the four digit hexidecimal address.

FREQ L: is the address of the low order (the fractional) part of the frequency register. The allowable values range from 0 to 255. The values are interpreted as a binary fraction with the decimal point to the extreme left side of the byte. This value is WRITE only and the two digit number in this column should be prefixed by the value \$C(S+1) to obtain the hexidecimal four digit address.

An Example

Suppose that you had your waveform table in memory page \$80 which corresponds to an address of \$8000. You would like to play the waveform on oscillator number A with a volume equal to three quarters of the full volume and at a frequency of 2.25 (which means that the waveform table pointer is incremented by 2.25 for each sample).

Here is what you should do:

1. Locate the line in the table for oscillator A.
2. Set the waveform table address register by writing the value \$80 into the location \$C(S+1)34.
3. Set the amplitude for the oscillator by writing the value \$C0 into the location \$C(S+1)35.
4. Set the frequency by writing the value \$02 into the location \$C(S+1)14 for the integral

part. Write the value $40 into the location $C(S+1)13 for the fractional part of the frequency.

In all of the above cases, except as noted for enabling and disabling the DMA, the READs and WRITEs may be done with PEEKs and POKEs from BASIC, or you may use LDA or STA to absolute addresses in 6502 machine language. Please be aware of the phantom read and write associated with the indexed, the indirect and the indexed indirect instructions of the 6502. These instructions have as side-effects additional and spurious reads and writes to the addressed memory. Normally the contents of the address will be okay, but the extra reference might confuse the register select logic. We prefer the use of absolute address instructions.

The Instrument Definition

Introduction

This section covers the software model used in the MusicSystem. The chapter started with a general introduction to the theory of additive synthesis. Several concepts essential for the electronic synthesis of music were introduced. The hardware section continued by discussing how the hardware model complements the theory of additive synthesis. Topics included the structure of the MusicSystem hardware and its operation, waveform table specifications, an I/O memory map and instructions for direct control of the hardware.

In this section, we continue to build on the previous sections by introducing the Instrument Definition. The Instrument Definition is essentially a software model which drives the hardware. The result is a powerful and flexible digital music synthesizer. This section will tie together many of the concepts introduced in the previous two sections. It is important to have read those sections before reading this section. Much of the information in this chapter will be helpful when you use the INSTRUMENT DEFINER program.

Several assumptions that were made during the design and implementation of the MusicSystem are discussed in this chapter. These assumptions are scattered throughout the text and they reflect decisions made while developing the MusicSystem. The information in this section will be helpful in understanding the operation of the MusicSystem. This section will also be helpful for those who wish to interface to our software, or for those who create their own software.

What Is An Instrument Definition?

An Instrument Definition is a large data structure. A data structure is a logical grouping of related data. The logical grouping depends on the data and the operations to be performed on the data. For the MusicSystem, the instrument definition is the data which identify the waveforms, and define the envelope and frequency history for a single note of a particular instrument. The instrument definition is a definition of timbre.

The instrument definition data structure is referenced by the name of the instrument definition file. Simply refer to a filename to obtain the pertinent data. Some instruments require only one set of waveform, envelope, and frequency history information, while others include multiple sets of information. The spectrum information is combined into a waveform and carried in the waveform table. Waveform tables are referred to by waveform name. The envelope data consists of the amplitude history which describes the attack, an exponential decay factor for the sustain, and a slope function (LOG or LIN) for the attack and decay slopes. The amount of time for both the attack and decay portions of the envelope are specified independently. Information about how the pitch of the note varies during attack is included in the frequency history.

This model of the instrument definition complements the theory of additive synthesis. In the first section we discussed analysis as a means of obtaining data for synthesis. Recall that the analysis was preformed upon a single steady note from the instrument. The same model is used for the instrument definition, which defines the timbre for a single note. Thus the software works with a note as a basic unit.

The Logical Oscillator

The concept of a logical oscillator is used throughout the instrument definition. The

digital oscillator was covered in the hardware section. The ditital oscillator is also called the physical oscillator. The logical oscillator is another data structure. It consists of one waveform, the amplitude and frequency histories and a few other variables. A logical oscillator is bound or assigned to a physical oscillator at "play-time". This binding to a physical oscillator is discussed later.

Consider an instrument definition where analysis has shown that there is one set of amplitude and frequency histories for the odd harmonics, and a different set of amplitude and frequency histories for the even harmonics. Remember that each logical oscillator has one waveform, and one set of amplitude and frequency histories. In this case you would combine the even harmonics into one waveform table and then describe the histories for that waveform. That defines the first logical oscillator. Next, you would combime the odd harmonics into a second waveform table, describe its histories, and thus define the second logical oscillator.

Now you then have an instrument definition which uses two logical oscillators, has two waveform files and two sets of amplitude and frequency histories. Observe that each different waveform or amplitude or frequency history requires a different logical oscillator. If you use three different waveforms to synthesize an instrument, you use three logical oscillators. Likewise, if the note you want to synthesize has three sets of curves which describe the amplitude and frequency histories, you will use three logical oscillators.

Think of the logical oscillator as the amplitude history, the frequency history and the waveform data that will be used by a digital oscillator to produce all or part of the sound of a single note played on the instrument. It does not matter which digital oscillator is used to implement the logical oscillator; the actual binding of the physical oscillator to the logical oscillator occurs later. The logical oscillator will produce all or part of the sound of the note depending on the number of logical oscillators used for one instrument definition.

The logical oscillator also includes a few more data items. Each logical oscillator has a relative weight, which determines its relative loudness. Additionally, the exponential decay factor describes the shape of the sustain portion of the envelope.

One Waveform Table Per Logical Oscillator

Each logical oscillator of an instrument definition has its own waveform. The waveform is a 256 byte table of amplitude values. It exists on the diskette as a binary file, and is used by the hardware. On the diskette, you must use the naming convention for waveform files, which is "WAVE." preceding an 11 (maximum) character waveform name. You may create your own waveforms by any method you choose and save them to use with the MusicSystem by following these instructions:

1. Organize 256 bytes of waveform data into a buffer in memory at address $XXXX. Select a waveform name, e.g., FOO
2. Assume that the table starts at address $XXXX, expressed in hexidecimal. Type this command from DOS:
   BSAVE WAVE.FOO,A$XXXX,L$100 return
3. The waveform is now ready to be incorporated into the definition of a logical oscillator.

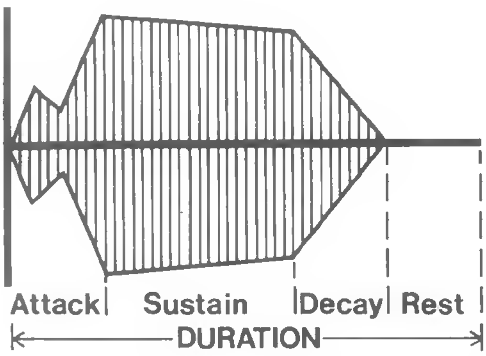
Waveform files created this way may be used anywhere a waveform file is used **except** for the WAVEMAKER portion of the INSTRUMENT DEFINER program. Only files that are created by WAVEMAKER may be used with WAVEMAKER. If you were to use a waveform file you created with the WAVEMAKER, then WAVEMAKER would behave as if it had a new waveform, i.e., the harmonics are set to zero.

You may use any source of data for creating a waveform file. An obvious application is collecting sound data for analysis by using the Mountain Computer A/D + D/A board. The data could be processed and used to create waveforms directly. The waveform data may also be the result of functional computations. The best advice is to experiment and have fun.

When you create your own waveform data, try to have at least one of the 256 samples at the maximum ($FF) or minimum ($01) range of waveform table values. This is not a strict requirement, but it does help even out the loudness of that waveform and logical oscillator. If this requirement is not met, the waveform will not be as loud as the others when played on the speakers.

One Amplitude History Per Logical Oscillator

Each logical oscillator used in the instrument definition has one amplitude history. The amplitude history describes the portion of the envelope called the attack. Remember that the logical oscillator describes the envelope for the duration of a single note. It is useful to think of the envelope as being made up of these parts: attack, sustain, decay and implicit rest. This illustration shows the relation:



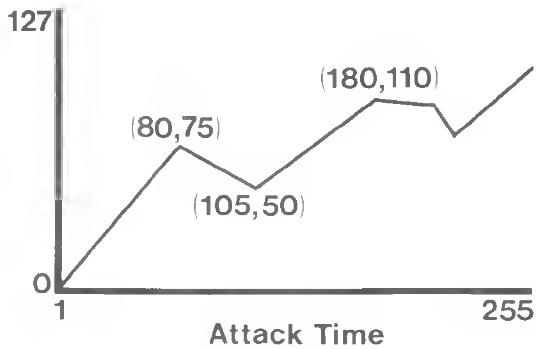
The envelope is a collection of information about the amplitude of a note. In the illustration, the horizontal axis is time. The envelope's time plus the implicit rest time equals the duration of one note. For the amplitude history, time is expressed as a number between 1 and 255. This time coordinate is relative time and is scaled as explained under Instrument Definition Globals. The vertical axis is the amplitude of the sound with a range of 0 to 127. The shaded area enclosed by the envelope represents the waveform.

The amplitude history describes how the amplitude of the sound starts from zero and builds up during the attack portion of the envelope. An implicit rest is appended to the end of the envelope. The implicit rest is approximated by a 64th rest and represents a short time between notes. The duration of the implicit rest is dependent on the tempo being used.

The Attack

While designing the MusicSystem, a few assumptions were made about the attack, sustain and decay portions of the envelope. Since studies have shown attack to be very important in the perception of timbre, you are able to specify the attack in detail. The INSTRUMENT DEFINER program allows for up to 15 points to specify the attack. Each point is an ordered pair of relative time and amplitude. Our model assumes that all attacks start with an amplitude of zero. The first point specifies the amplitude at some relative time. The second point specifies a different relative time and amplitude coordinate. The ordered pairs are plotted on a graph, and the points are connected by a straight line. These straight lines are the "slopes" of the attack. In this manner, the slopes (the attack portion of the envelope) are specified by a set of ordered pairs.

You express the attack as a set of slopes by entering the ordered pairs in the INSTRUMENT DEFINER program. Below is the attack portion of the envelope drawn in detail. As you can see, the attack is made up of slopes specified by the ordered pairs. The first number in the ordered pair is a time coordinate expressed in relative time units and the second number is the amplitude. The graph shows the maximum and minimum values for each number in an ordered pair.



Notice that the slopes in the illustration are straight lines. If you display the amplitude history, the slopes are always straight lines. The LOG/LIN global parameter effects the interpretation of the attack slopes. LOG/LIN selects the amplitude scale used in the display. If LIN is selected, the amplitude of the attack will change linearly. If LOG is selected, the amplitude of the attack will change logarithmically. That is because a straight line on a LOG graph is the same as a logarithmic curve on a LIN graph.

In most cases, the limit of 15 ordered pairs will not be a problem. Most attacks can be

described in fewer ordered pairs. There is a possible loss of information about the slopes of the attack when the duration of the note is short and the number of ordered pairs is great. Another reason for loss of information might be too many ordered pairs in too short of a global attack time. The global attack time describes the number of time units for the attack portion of the envelope for all logical oscillators. Thus the attacks of each different logical oscillator in one instrument definition start and end at the same time. The relations among these parameters are covered in detail under the heading Instrument Definition Globals.

The Sustain

The sustain of the envelope is expressed by an exponential decay function. You may specify the exponential decay factor of the sustain with the INSTRUMENT DEFINER program. The sustain's decay factor is the number of milliseconds between decays of the sustain amplitude. At each one of these times, the amplitude will become one half of its previous value. Thus the sustain decay factor is the half-life of the sustain expressed in milliseconds.

If the exponential decay factor were equal to 2, the slope of the sustain would be steep, since the amplitude would be half its previous value each 2 milliseconds. If the exponential decay factor were the maximum value of 2048, for all practical purposes there would be no decay during sustain. You can specify the slope of the sustain with this factor. The duration in time units of the sustain is variable and depends on the duration of the note and the tempo.

Time Units

Time units are based on the interrupt rate of the MusicSystem hardware which provides synchronization for the software. The MusicSystem is shipped with an interrupt rate of one interrupt per 8 milliseconds. The timing of all events (rests, the duration of notes, the attack time) is based on counting these periodic interrupts. The time unit is also called an interrupt and it is always 8 milliseconds. Therefore 5 time units is the same as 5 interrupt counts.

Since the smallest time increment for the MusicSystem is 8 ms., you cannot specify events separated by less than 8 ms. However, under certain conditions, the minimum separation between events may go up to 16 ms. This is covered in the section on Time Resolution.

Time units are used to time notes, or to time the ordered pairs in the amplitude history of a logical oscillator. The amplitude history, the frequency history, the sustain decay and the decay all use interrupt counts as the basic time unit.

Time Resolution

The MusicSystem hardware generates one interrupt per 8 ms. The MusicSystem software uses these interrupts to time events. Under most conditions, the minimum time between events is 8 ms. This minimum time between events is called the time resolution. The MusicSystem software does all its timekeeping with these 8 ms. time units, and it cannot distinguish any durations shorter than 8 ms. However, under extreme conditions, the MusicSystem will resort to a time resolution of 16 ms. This happens when there are many parts in a composition, and many logical oscillators are being used. Under these conditions, 8 ms. is not enough time to do everything. When this happens, the MusicSystem will increase the time resolution to 16 ms.

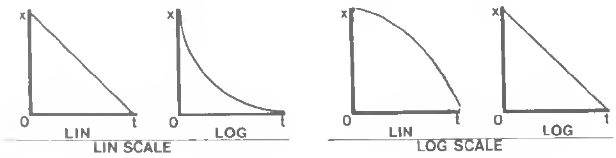
This will directly effect the amplitude history. Consider the case where you want to space the ordered pairs of the amplitude history 8 milliseconds from each other. There is no problem until the MusicSystem begins to emulate 16 millisecond interrupts. When that happens, some of the ordered pairs will be lost because of time resolution. The point here is to be careful and don't get caught by time resolution. Play it safe and don't specify events to occur any closer to each other than 16 milliseconds. For this reason, the attack time is expressed as an even number of time units.

One implication of time resolution applies to the amplitude history and the global attack time. When you specify a history, you use relative time units which are adjusted to real time units when the instrument definition is generated. Suppose you define a history using the relative times 1 through 80. If you specify that the attack time is 10 (time units), this will be 80 ms. (Each time unit is 8 ms., 10 times 8 is 80 ms.) Thus each relative time unit is

one millisecond. If you specified amplitude history ordered pairs with the relative time being integral multiples of 16 (e.g., 16, 32, 48, 64, etc.), everything would be fine. If you specified an ordered pair at relative time 4, that would exceed the resolution of 16 ms., and that ordered pair would be lost. The lesson here is be careful about time resolution. Do not specify events any closer than 16 milliseconds.

The Decay

The decay portion of the envelope is also an important part of the timbre. The decay can best be modeled by either a linear function or a logarithmic function. In either case, the decay portion of the envelope starts at whatever value of amplitude the sustain reached at the end of sustain time. The amplitude proceeds to zero amplitude either linearly or logarthmically in a specified amount of time.



The above illustration shows the shape of the decay curves for a logarthmic decay (LOG) and a linear decay (LIN). X is the value of the amplitude at the beginning of decay time. The amplitude scale (y–axis) is linear for the two graphs on the left. The amplitude scale is logarthmic for the two graphs on the right. Time "t" equals the global decay time. Notice that a LOG decay appears as a straight line on the graph with a log scale.

The slope of the decay depends upon the envelope's amplitude at the beginning of decay time and the amount of time specified by the decay time. The decay time is global for the instrument definition, that is, all logical oscillators in one instrument definition have the same decay time. Decays start and end at the same time for the logical oscillators of an instrument definition.

After the decay comes an implicit rest. The rest is always the same duration as a 1/64th rest. The actual length in time of the rest is determined by the tempo.

Duration And The Envelope

The attack and decay times are global variables for the instrument definition. The implicit rest is of fixed duration. The duration of the sustain is variable and can be calculated with this formula:

sustain = duration – attact –decay – implicit rest

The duration of a note is implemented by counting a certain number of interrupts. The tempo and the duration of the note being played will determine how many interrupts must be counted for a note. Of the note's duration, a certain specified amount of time is for the attack, and another specified time is for the decay. The number of time units (or interrupts) for the rest depends on the tempo.

Together the attack, decay and rest will account for some portion of the duration. If there is any time left after subtracting the needs of the attack, decay and rest, it is used as the sustain time. It is possible to specify an attack time plus a decay time that is greater than the duration of a note. This implies that there is no time left for the sustain. That case is covered under the heading of The Eating Algorithm.

The Importance Of The Attack

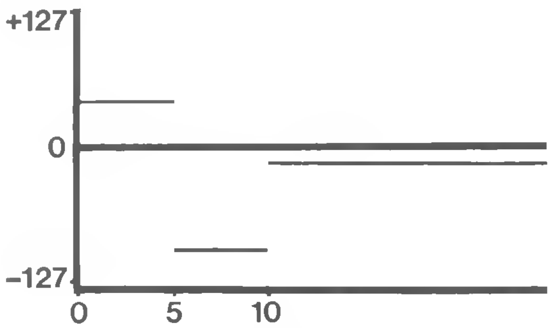
When specifying the amplitude history of an instrument you wish to synthesize, you will find the attack portion of the envelope contributes greatly to your sense of timbre. Because the attack is critical to a good instrument definition, the software model allows more points than are usually necessary to specify the slopes of the attack. You may specify up to 15 ordered pairs in the amplitude history.

If you use many ordered pairs to specify the attack, and the global attack time is small, it might not be possible to represent all the ordered pairs with the available time resolution. This results in an attack different from the one you specified, and sometimes the wrong timbre. The Instrument Definition Globals section has a detailed treatment of this consideration. Our advice is to let your ear be the final judge.

One Frequency History Per Logical Oscillator

Recall that a waveform is synthesized from a sine wave and its harmonics. The distinction between a harmonic and a partial was made because analysis reveals that the "harmonics" are not exact integral multiples of the fundamental frequency. The frequency history is a mechanism which specifies how the pitch of a note is to vary during the attack time. Frequency history also contributes greatly to the timbre of the instrument. Studies have shown that the frequency of a note varies during the attack, and usually stabilizes after the attack. This is the assumption made for the instrument definition.

Each musical pitch has a particular frequency. When the software plays a note, the pitch of the note determines the value loaded into the frequency register of a digital oscillator. The format of the number in the frequency register is explained in the hardware model. The frequency history of the instrument definition specifies a positive or negative value to be added to the frequency register at particular times. In other words, the base frequency value is adjusted by the frequency history. Below is an illustration of a typical frequency history.



The horizontal scale is time, ranging from 1 to 255 relative time units. These relative time units are scaled in a way similar to the amplitude history's time units. Remember that the frequency history spans the same time as the amplitude history. The vertical scale is plus 127 units at the top, 0 units in the middle and minus 127 units at the bottom. Each unit of the vertical scale represents one unit of frequency resolution, namely 0.5 Hz. The frequency history is drawn as a discontinuous function in time.

In the above illustration, at the beginning of the attack, the base frequency of the note is increased by 50 units or 25 Hz. Then at time 5, the base frequency of the note is decreased by 100 units or 50 Hz. Then at time 10, the base frequency is decreased by 10 units or 5 Hz. The frequency remains unchanged for the rest of the frequency history. The frequency history ends at the same time that the attack ends. Whatever value is in the frequency register at the end of attack time is the fixed frequency for the rest of the duration of the note.

The frequency history is an essential part of the timbre of an instrument definition. When collecting data for frequency history from published data, you will have to make a "best-fit" estimate for the frequency history ordered pairs. When you enter frequency history information into the INSTRUMENT DEFINER program, you will be entering ordered pairs. The first number in the pair is a time coordinate between 1 and 255, and the second number is between plus and minus 127 and represents the difference in the frequency of the note.

The frequency history will allow for up to 15 ordered pairs. In common practice, the frequency history can be specified in fewer ordered pairs. You will have to experiment to obtain the best results. The problems of resolution and loss of information for the frequency history will occur under the same conditions as for the amplitude history. These conditions are too many ordered pairs for a given attack time. Remember that the global attack time also specifies the time for frequency histories. This consideration is discussed in detail later.

One Oscillator Weight Per Logical Oscillator

In the instrument definition, each logical oscillator has a relative weight factor. The factor represents the relative weight of the logical oscillator's amplitude. When an amplitude history has been specified, there will be some maximum amplitude value that was used. This maximum value is scaled up to a value of 127. This scaling process might change some of the slope information in the amplitude history. The resulting amplitude history is then scaled by the logical oscillator weight factor.

For example, if the weight factor is 100, then the maximum point of the amplitude history is scaled to 127 but not adjusted any further. If the weight factor is 50, the maximum point of the amplitude history is scaled to 127 first, and then scaled to half that size. This second scaling can change some of the slope information again. This factor is used to adjust the relative loudness of the logical oscillator in the instrument definition.

The actual formulas and procedures for calculating this scaling follow: First, for the amplitude history entered, the maximum amplitude used is found and called amp_{max}. Then for each ordered pair specified, the amplitude at that ordered pair (amp_i) is assigned the value of that ordered pair's amplitude multiplied by 127 and then divided by the maximum amplitude.

$$amp_i = amp_i * 127 / amp_{max}$$

This first calculation scales the amplitude history so that there is one point at the maximum amplitude. Next, the ordered pair's new amplitudes are multiplied by the weight factor, then divided by 100 and assigned into the ordered pairs. These calculations are internal to the software, yet are presented here so that you are aware of the mechanism used. The second formula uses the weight factor that you specify.

$$amp_i = amp_i * weight / 100$$

One Exponential Decay Factor Per Logical Oscillator

The sustain has been discussed already. Recall that the length of time of the sustain is variable and that you may specify the slope of the sustain. In our model of the instrument definition, we assume that the sustain can be modeled by an exponential decay. Many natural events have been observed to follow exponential decays. Essential to the concept of exponential decay is the half-life. The half-life is the amount of time it takes for a given quantity to be reduced to one half its original quantity. In this context, the quantity is the amplitude of the envelope.

The exponential decay factor is expressed in milliseconds, not time units, and is the number of milliseconds it takes for the sustain amplitude to reach half its previous value. The sustain value starts at the amplitude reached at the end of attack time. Since the duration of the sustain is variable, there is no way to predict the value of the amplitude at the end of the sustain time for note after note. By specifying the exponential decay factor, you specify the slope (rate of decay) of the sustain.

Instrument Definition Globals

The instrument definition is made of several parts. The amplitude history, the frequency history, the waveform, the wieght of the logical oscillator, and the exponential decay have been discussed. They must be specified for each logical oscillator of an instrument definition. It is possible for several logical oscillators to share one waveform, for example, but each logical oscillator will have its own weight, decay factor, amplitude and frequency history.

Part of the instrument definition applies to all the logical oscillators in the instrument definition. This part of the definition contains two global values. One value specifies the amount of time to spend on the attack portion of the envelope. The other value is the amount of time to spend on the decay portion of the envelope. Both the attack time and the decay time are expressed in interrupt counts (time units). Therefore an attack time of 2 would be 2 times 8 ms. or 16 milliseconds. The time must be an integer, and if odd, it will be rounded up to the next even integer.

Both the attack and the decay globals may be set with the INSTRUMENT DEFINER program. Recall that the frequency history spans the same time the amplitude history does. That amount of time is specified by the attack time. The decay time specifies how much time is to be spent in the decay portion of the envelope. Recall that the decay will start from the amplitude at the end of sustain.

The Instrument Definition allows you to specify either a LOGarithmic or a LINear function for the attack and decay slopes. When you display the amplitude history, the slopes are displayed as straight lines. The vertical axis is either a LOG scale or a LIN scale. With LIN selected, the amplitude history slopes and the decay slope are linear functions. When LOG is selected, the amplitude history slopes and the decay slopes are logarithmic functions.

Global Time And History Considerations

When you specify either the frequency history or the amplitude history for a logical oscillator, you use relative time units in the ordered pairs. These relative time units are not milliseconds, nor are they time units (interrupt counts). The relative time units are just that—relative. The relative units have no meaning until a global attack time is specified. The attack time applies to both types of histories.

The relative time units have meaning only when a global attack time is specified. The attack time is in interrupt counts (time units) and therefore it must be an even integer because of time resolution. When you specify a history, you may use any time coordinates you wish. There will be some maximum time unit that you use. That maximum will be less than or equal to 255. You may specify the whole history using relative times 1 through 10, or you may use from 1 to 200; the choice is yours to make.

Suppose that you use up to 100 on the relative time scale. Next you assign an attack time of 10 interrupt counts. The maximum relative time is scaled to be the attack time; in this case 80 ms. (10 * 8 ms. = 80ms.). Every ordered pair in the history is scaled by the same factor. If there was an ordered pair at relative time 20, it would occur at the time unit of 2. By knowing the attack time, and the maximum relative time, you can compute the time of occurrence for an ordered pair. The formula is:

$$\text{time} = \text{time}_b * \text{attacktime} / \text{maxtime}_h$$

Attacktime is the global attack time expressed in time units, maxtime_h is the highest relative time from the history, time_b is the relative time of the ordered pair, and time is when the event will occur expressed in time units from the beginning of the attack. If the result of the formula is not an even integer, it will be rounded up to the next even integer.

Because the time resolution of the MusicSystem is two time units or interrupts, the MusicSystem software cannot perform two events A and B sequentially if B is to occur less than 16 milliseconds after A. Two events will be separated by 16 milliseconds or integral multiples of 16 ms.

Suppose that you specify an amplitude history and you use relative times 1 through 100. Suppose further that you used integral multiples of 10 for all the relative time coordinates. Next, you specify the attack time as being 10 interrupt counts. This means that relative time 10 is one interrupt count, relative time 20 is two interrupt counts, etc. Recall that the time resolution is two interrupt counts. This means that the ordered pairs occurring at relative times 10, 30, 50, 70, and 90 exceed the time resolution. Recall that you cannot specify events any closer than two interrupt counts. Thus events that should occur at one, three, five, etc., interrupt counts will not occur.

How does this effect the shape of the attack of the envelope? It is possible that the actual shape of the attack will change. In the above example, all events which were scheduled to occur at odd interrupt counts did not occur. The events scheduled to occur at even interrupt counts will occur just as specified. But the slopes of the attack will change. Suppose there was an ordered pair at relative time 10 and another at relative time 20 in our example. The actual slope produced by the MusicSystem will be a slope from time 0 at 0 amplitude to time 20 at whatever amplitude you specified at time 20. The ordered pair at time 10 is ignored because it exceeds the time resolution of two interrupt counts.

The slope you hear will be different from the slope you specified. Some of the ordered pairs might be lost because of the scaling of the relative time scale, and you should be careful when specifying the history and the attack time. It is not important to know exactly how the slopes are changed. It is mentioned here so that you will have a better idea of the mechanics of the MusicSystem. What is important is how it sounds. The theory should give you an idea of where to start, and your ear should guide you in "fine-tuning" the instrument definition to suit your taste.

In summary, you should choose with care the relative times in your history definition, and consider time resolution when specifying an attack time. You may also consider it from a different point of view. Specify the attack time first, and then assign the relative times so that no information is lost due to time resolution. Always listen to the sound and don't be afraid to experiment.

The Instrument Definition Program As A Tool

The instrument definition is a large and complex data structure. There is much to be considered when making an instrument definition. We have prepared a program for the MusicSystem to assist you in the development of instrument definitions. This is the INSTRUMENT DEFINER program, covered in Chapter 8.

All the topics mentioned so far, the logical oscillator, the waveform, the amplitude history, the frequency history, the exponential decay factor, the oscillator weight, and the attack and decay times may be specified, set and modified by the program. You should read Chapter 8 to gain an understanding of how to run the program. This chapter provides some background and theory which will be helpful when using the program.

The Instrument Definition Environment

The Instrument Definition describes the timbre of an instrument. The data used to create the definition is usually obtained from analysis of a single note played on the instrument. This will yield a good approximation of the instrument, but only over a certain octave range. Many instruments have a different spectrum and different amplitude histories when played in a different octave. Since analysis is performed on one note, the timbre defined by that analysis will be limited to a range of notes near the pitch used in analysis. This range varies for different instruments.

When you use the MusicSystem, you will find that there is a range of pitches that can be produced by an instrument definition. For example, a tuba has a range of pitches it can play. A very high note played on a tuba will sound very strange indeed! At the end of this chapter is a chart of instruments included with the MusicSystem. In this chart, you will find a recommendation for the octave range of the instrument. We encourage you to experiment and let your ear guide you.

The instrument definition lives in the environment of your Apple. There are several programs in this environment, and some of them are described below. This environment also includes some data structures, and some practical specifications. The purpose here is not to describe each and every program in detail, but to give you an idea of the other pieces which make up the MusicSystem software. The data structure of the composition file is discussed. The eating algorithm of the producer and consumer is examined in detail because it affects the envelope of the note.

The Comp File

One of the largest data structures used in the MusicSystem is the composition file, also called the COMP file. This file is created with the MusicSystem Editor, and it represents the sheet music of the song to be played. It is possible to play the composition with any instrument.

The COMP file is produced and modified by the Editor program. The COMP file is then processed by the Player program. The Player transforms the COMP file into a format suitable for the Producer and Consumer. The Producer and Consumer together are responsible for driving the hardware from the data which represents the song.

The Producer And The Consumer

There are two programs in the MusicSystem whose relationship is so close that it influenced their names. The Producer is a program that receives the output of the Player program. The Producer takes that information and combines it with the instrument definitions to produce an output queue. This output queue contains information about events to be performed by the hardware. The queue is time ordered, that is, earlier events come before later events in the queue.

The Consumer takes this time ordered queue as input. The consumer controls the hardware according to the instructions in the Producer-Consumer queue. Both the Producer and Consumer are controlled by a Supervisor program. The Supervisor decides when to run the Producer or the Consumer. Together the Producer, the Consumer, and the Supervisor share the CPU's resources when the MusicSystem is actually playing music. Those three programs, along with the Player's output comprise the "play-time" environment of the MusicSystem.

Interrupt Driven

The Supervisor program is an interrupt driven program. The MusicSystem hardware generates interrupts at a jumper selectable rate. The MusicSystem software uses the 8 ms. interrupt rate. This means that one interrupt is generated every 8 ms. Since the interrupt rate is derived from the Apple's timing crystal, it is not exactly 8 ms. between interrupts.

If a program is interrupt driven it means that every time an interrupt occurs the interrupt driven program gains control of the computer, i.e., it starts to run. In this case, when an interrupt occurs, the supervisor program starts to run. The supervisor will update its clock and then check a few things. The Supervisor will then run the Consumer which will process all the events for that time. Next, the Producer is run so that the queue may be kept full. Control returns to the Supervisor, and then to the interrupted program.

The Eating Algorithm

There will be times when you want to specify a large attack time and a large decay time for an instrument definition. Suppose you set the attack and decay times to large values, say 100 each. Remember that 100 is the number of interrupts to count, so the attack and decay would be 800 ms. each. Suppose that with such an instrument definition you tried to play eighth notes at a tempo of 140 quarter beats per minute. The problem arises when you consider that for a note of that duration at that tempo, there simply is not 1.6 seconds to spend on the attack and decay both. The note's duration and the tempo alone imply almost 5 notes per second.

What happens to the envelope during such a situation? It is clear that there is not enough time in the duration of a single note for the attack time and the decay time specified. To say nothing of the implicit rest and the sustain. There is an algorithm programmed into the MusicSystem software which describes what to do in such a case. This is called the Eating Algorithm because it decides which parts of the envelope to "eat" or throw away.

The Eating Algortihm is used when the duration of the note to be played at the current tempo exceeds the minimum duration of the envelope. The minimum duration of an envelope is the attack time plus the decay time plus the implicit rest time. Recall that we said that the sustain time is variable, and equals the time remaining from the subtraction of the attack, decay and rest times from the duration of the note. If the duration of a note is exactly equal to the sum of the attack, decay, and implicit rest times, there will be no sustain. This situation is okay and does not require the Eating Algorithm. But if the sustain has been eliminated, and there still is not enough time in the note's duration for the envelope, the Eating Algorithm is used to fit the envelope into the available time.

If the note's duration minus the attack time minus the decay time minus the implicit rest time is less than zero, then enough of the implicit rest is thrown away to make the envelope fit the note's duration. Only enough of the implicit rest is thrown away to make the envelope match the duration.

How long or how many counts is a note? The number of interrupts counted to time a duration is based on this formula:

$$IC = (122 \times 15) / (4 \times Q)$$

Q is the tempo expressed in quarter beats per minute. The value of 122 represents the number of interrupts per second, and the number 15 is a constant. The result, IC, is the number of interrupts to count for the duration of a sixty-fourth note. To find how many time units a quarter note at a tempo of 72 takes, you first calculate IC:

$$IC = (122 \times 15) / (4 \times 72) = 1830 / 288 = 6.35$$

It takes 6.35 interrupts for one sixty-fourth note. The quarter note is 16 times longer than the sixty-fourth note, so 16 * 6.35 is 101.6 counts for a quarter note. The duration calculations are carried out with accuracy, but the final count must be an even integer because of time resolution. Now you know how to calculate the number of interrupts used to time note durations. The same units are used to specify the attack time.

If throwing away the implicit rest was not enough to make the envelope fit the duration, the Eating Algorithm continues. At this point the sustain is gone, and the implicit rest is gone. All that remains is the attack and the decay. You get to this point in the Eating Algorithm if the attack and decay together are greater than the duration of the note. The next part of the envelope to be eaten is the decay.

The Eating Algorithm throws away all or part of the decay so that the envelope will fit the duration. This will affect the sound you hear on the speakers. If the next musical event after that envelope is a rest, the decay will behave as if it had enough time to decay smoothly to zero. If the next musical event is something other than a rest, there will be a small discontinuity in the amplitude, possibly resulting in an audible click.

In the worst case, if the Eating Algorithm throws away the decay and there is still not enough time in the duration for the attack, good luck! If the duration of the note is exactly equal to the attack time, then the whole attack portion of the envelope is played. If the duration is less than the attack time, some of the attack is thrown away. If the duration is less than the attack time and greater than 16 time units, then the attack becomes a linear slope from zero to the maximum amplitude of the amplitude history. The maximum amplitude occurs at 16 time units (from the beginning of the envelope) The rest of the duration is treated like a sustain, that is, the exponential decay factor determines the amplitude of the envelope.

If the duration is less than the attack time and less than 16 time units, then the attack becomes a straight line slope from zero amplitude, zero time to maximum amplitude (of that history) at duration time. If the duration is ever less than 16 time units, the frequency history is ignored, and the note is played at the base frequency of the pitch.

The Eating Algorithm is not the best solution to this problem. You can see that important parts of the envelope are thrown away, and the slope information of the attack can become very distorted. The Eating Algorithm will sacrifice timbre for tempo and duration. During implementation, it was judged better to throw away timbre information than to throw away tempo and duration integrity.

There are a few implications to all of this. If you are going to play a composition at a very fast tempo, you would do well to define the instrument with short attack and decay times. With this method the timbre will not suffer at higher tempos. If you want total control over the envelope define the entire envelope into the amplitude history. This will place the entire envelope into the attack portion of our envelope model. As long as the duration is not less than the attack time, the total definition will remain. If the duration is longer than the attack time, the remaining time is silence, assuming your envelope ends at zero amplitude.

Allocation Of Oscillators

In the instrument definition, logical oscillators are assigned many characteristics. These logical oscillators are not assigned to one of the physical oscillators of the MusicSystem hardware until later. Complex timbres require more logical oscillators than simple timbres. Eventually each logical oscillator is assigned to a physical oscillator, so you can play many simple instruments, or a few complex instruments.

The MusicSystem boards contain 16 physical oscillators to produce 16 separate elements of the two audio signals. Eight physical oscillators are dedicated to the right channel output and eight physical oscillators are dedicated to the left channel output. The number of parts (or voices) that can be used in a composition is determined by the way the instrument is defined and the way it is output to the physical oscillators.

For example, an instrument has been defined to approximate the sound of a bass clarinet, and has been named "CLARINET." A very rough approximation of a bass clarinet could be produced with a single oscillator, but it would sound as much like a tuba as it does a bass clarinet. More typically, an instrument like a bass clarinet requires three physical oscillators to produce a sound that is distinctly similar to the natural sound of a bass clarinet.

A single part in a composition (for example, bass clarinet) will use three oscillators. If the part is designated as coming from a single channel (left or right), then 3 physical oscillators (again left or right) will be used. If the same part were designated as stereo (sound coming from both channels), then 3 oscillators from the left, and three oscillators from the right are used. However, if the entire composition is designated as monophonic, 3 arbitrarily chosen oscillators will be used to produce that part. For example:

If a composition includes 5 parts, each using 3 oscillators, you will be able to use a differing

number of these parts depending upon how the parts and the composition is defined spatially. Remember that each channel has 8 oscillators; thus, each side can contain instruments in "sets of three oscillators," up to a total of 8 per side.

If specifying left or right locations, you could include 4 parts of three-oscillator instruments. If specifying stereo, you can only include 2 parts of three-oscillator instruments. But if you specify mono for the entire composition, you can use 5 parts of three-oscillator instruments.

Use the chart at the end of this chapter to determine the number of oscillators required for each instrument and be aware of the effect of defining a part as coming from both speakers (in the Music Editor) or defining the composition as monophonic (in the Music Player).

How To Determine Maximum Polyphony Size

The number of notes in a chord determines the number of oscillators that can be used in any composition. The Music Player displays information about a composition before actually playing it, including the largest number of notes in a chord in each part. For example:

PART #	NAME	INSTNAME	SPKR
1 (1)	PART 1	BRASS	RIGHT
2 (1)	PART 2	ORGAN	LEFT

In the column labeled "PART #," two sets of numbers are given: the left most is a sequentially assigned part number, the number given within the parentheses is the "maximum polyphony size" for that part. Simply put, maximum polyphony size is equal to the number of notes in the largest chord in each part. Each part in a composition could have a different maximum polyphony size; similarly each instrument in a composition can require a different number of logical oscillators to produce its particular sounds. To determine the number of physical oscillators that will be needed to produce each part, use this formula:

# of physical oscillators = # of logical oscillators * max polyphony size

The number of logical oscillators an instrument requires remains constant; refer to the Chart of Instrument Definitions for that value. The value of the maximum polyphony size will vary from part to part and is always displayed by the Music Player. For example:

If instrument "ORGAN" requires 2 logical oscillators, and the maximum polyphony size for a part that uses "ORGAN" is 4 (that is, there is at least one chord with four notes in it), the total usage of physical oscillators for that part is 8.

The only time you need be concerned with maximum polyphony size is when a composition is written using chords. It has no effect on parts written using single notes, as the polyphony size of notes is 1.

Waveform Creation

You may use the INSTRUMENT DEFINER program to create and modify waveforms for the MusicSystem. It is also possible to use any other source to create your own waveforms. The basic requirement for a waveform table is 256 consecutive bytes of amplitude data. The waveform table represents 256 amplitude samples of the waveform. The amplitude values may range from $01 to $FF. $01 is the most negative amplitude, $80 is zero amplitude, and $FF is the most positive amplitude. The waveform table must begin on a page boundary.

You may write programs which will create waveform tables and then save them to disk. Be sure to use the naming convention outlined above. The programs may calculate the waveform or read the game paddles to make entries in the waveform table. You could use a Mountain Computer A/D + D/A board and collect 256 samples for your waveform. If you wish to experiment with aperiodic instruments like drums, then a waveform table of noise would be helpful. Simply fill a waveform table with 256 random values to create a noise waveform. Remember that any 256 bytes of data can be used for a waveform.

An Example Program For Waveform Creation

We encourage you to create your own waveforms by any means possible. Included here are two simple APPLESOFT programs that may be used to create waveforms. You may use these programs as they are listed or you may expand them.

The first program is a general purpose waveform generator. It will prompt you for the number of harmonics. The maximum number of harmonics is 20. Next, you are prompted for the harmonic's relative amplitude weights which should be a number between 0 and 100. The program prompts you for the phase difference (in radians) of each harmonic. After the program has calculated the waveform, you are prompted for the name of the waveform. The file will be named exactly as you specify. If you wish to use the waveform, specify "WAVE." as the first five characters of the name.

```
110 REM GENERAL PURPOSE WAVEFORM PROGRAM
120 HIMEM:32000
140 BC = 33024
150 PI = 3.14159265
160 DIM FS(2,20)
170 DIM WT(256)
180 P2 = PI * 2
200 GOTO 1500
900 REM ROUTINE TO BUILD A WAVE TABLE
910 REM WITH HARMONICS AND PHASE DIFFERENCES
930 C = 0
970 FOR I = 0 TO 255
980 S = 0
990 FOR K = 1 TO N
1000 S = S + SIN ((K * P2 * I / 256) + FS(2,K)) * FS(1,K)
1010 NEXT K
1020 WT(I + 1) = S
1022 IF ABS (S) >C THEN C = ABS (S)
1030 NEXT I
1040 FOR I = 1 TO 256
1050 POKE BC + I - 1, INT (WT(I) * 127 / C + 128.5)
1060 NEXT I
1070 RETURN
1500 FOR I = 1 TO 20
1510 FS(1,I) = 0
1520 FS(2,I) = 0
1530 NEXT I
1800 PRINT "ENTER NUMBER OF HARMONICS"
1810 INPUT N
1900 PRINT "ENTER HARMONIC AMPLITUDE WEIGHTS (0 - 100)"
1910 FOR J = 1 TO N
1920 INPUT FS(1,J)
1930 NEXT J
1950 PRINT "ENTER PHASE DIFFERENCES"
1960 FOR J = 1 TO N
1970 INPUT FS(2,J)
1980 NEXT J
2100 GOSUB 900
2200 D$ = CHR$ (4)
2210 INPUT "ENTER WAVEFORM FILE NAME - ";N$
2220 PRINT
2230 PRINT D$;"BSAVE ";N$;",A";BC;",L256"
2240 END
```

The second program is much simpler, it creates a square wave and saves it into a file. Similar programs could be written to make ramp waves, sawtooth waves, pulse waves or random waves.

```
100 REM SQUARE WAVE PROGRAM
110 HIMEM:32000
120 AD = 33024
130 FOR I = 0 TO 127
140 POKE AD + I,255
150 POKE AD + 128 + I,1
160 NEXT I
170 D$ = CHR$ (4)
180 PRINT D$;"BSAVE WAVE.SQUARE,A";AD;",L256"
190 END
```

A Chart Of Instrument Definitions

Name	Number of Oscillators	Octave Range	Waveform Files	Description
ORGAN	2	0 - 7	IDEF.ORGAN WAVE.ORGAN1 WAVE.ORGAN2	full keyboard range
CLARINET WOODWIND	3	2 - 5	IDEF.CLARINET WAVE.CL1 WAVE.CL2 WAVE.CL3	sounds like woodwind from G2 to G5
BASS	1	1 - 2	IDEF.BASS WAVE.BASS	sounds good up to G3
GONG	2	1 - 5	IDEF.GONG WAVE.GONG	see note 1.
DRUM	2	1 - 3	IDEF.DRUM WAVE.SINE1 WAVE.DRUM2	good bass drum from G1 to G3 see note 2.
WOODBLOCK	2	1 - 4	IDEF. WOODDRUM WAVE.SINE1 WAVE.DRUM2	see note 3.
BRASS	3	2 - 4	IDEF.BRASS WAVE.BRASS1 WAVE.BRASS2 WAVE.BRASS3	see note 4.
PIANOLOW	1	1 - 3	IDEF.PIANOLOW WAVE.SAWTOOTH12	G1 to G3
PIANOMID	2	3 - 5	IDEF.PIANOMID WAVE.SAWTOOTH16 WAVE.SINE2	C3 to C5
PIANOHIGH	2	4 - 6	IDEF.PIANOHIGH WAVE.SINE1 WAVE.SAWTOOTH12	G4 to G6 and beyond
CYMBALS	2	2	IDEF.CYMBALS WAVE.NOISE WAVE.UGLY	B2 or C3
CLAVICHORD	2	0 - 7	IDEF.CLAVICHORD WAVE.CLAVICHORD WAVE.NOISE	full range
			WAVE.NOISE	

NOTES

1. The GONG instrument serves as two instruments, based on the octaves used. For octaves 1 - 3, the result sounds like a chime; for octaves 4 - 5 the result is more bell-like.

2. The DRUM instrument provides a good bass drum sound over G1 to G3. An excellent, non-tonal sound is produced at C0.

3. The WOOD BLOCK provides good bass sound from C1 to C2, and a good "wood box" instrument sound from C3 to G4.

4. The BRASS produces a tuba-like sound from G1 to C3, and a horn-like sound from C3 to C5.

A Chart of Pitch Values

This chart is arranged into columns which represent the octaves. Each column starts at the top with the C of that octave, and reads down the column to B. Next to each note is the pitch value for that note. These same values are used in the transpose specification of the INSTRUMENT DEFINER.

CHART OF PITCH VALUES
OCTAVE NUMBER

0	1	2	3	4	5	6	7
-----	-----	-----	-----	-----	-----	-----	-----
C -0	C -12	C -24	C -36	C -48	C -60	C -72	C -84
C#-1	C#-13	C#-25	C#-37	C#-49	C#-61	C#-73	C#-85
D -2	D -14	D -26	D -38	D -50	D -62	D -74	D -86
D#-3	D#-15	D#-27	D#-39	D#-51	D#-63	D#-75	D#-87
E -4	E -16	E -28	E -40	E -52	E -64	E -76	E -88
F -5	F -17	F -29	F -41	F -53	F -65	F -77	F -89
F#-6	F#-18	F#-30	F#-42	F#-54	F#-66	F#-78	F#-90
G -7	G -19	G -31	G -43	G -55	G -67	G -79	G -91
G#-8	G#-20	G#-32	G#-44	G#-56	G#-68	G#-80	G#-92
A -9	A -21	A -33	A -45	A -57	A -69	A -81	A -93
A#-10	A#-22	A#-34	A#-46	A#-58	A#-70	A#-82	A#-94
B -11	B -23	B -35	B -47	B -59	B -71	B -83	B -95

Appendix A
Glossary of Musical Terms

This brief glossary provides a quick reference to some musical terms used in describing the MusicSystem. It is not an authoritative source of information, but rather a quick reference of terms particular to this product. Refer to any current dictionary of music for further information and explanations.

A, B, C

accent — used to stress a note or chord; two types of accents are used in the Music Editor, regular and dynamic

accidentals — any sharp, flat or natural which is not given in the key signature

allegro — a tempo marking meaning "fast", from the Italian word for "happy"

andante — medium tempo. About 60 beats per minute, or the tempo of a slow steady walk

asterisks — used in the Music Editor to display the occurrence of special musical events which have no standard notation in music. Also referred to as "stars".

bar lines — a vertical line through the staff dividing one measure from the next

chords — a simultaneous combination of three or more notes of different pitch. Moving from one chord to another is "harmony".

clef — the symbol placed at the beginning of a musical score to indicate the sound range

composition — any musical work

D, E, F, G

dotted notes and rests — a dot on a note or rest increases the duration of the note or rest by one-half; e.g., a dotted half-note equals a half note plus a quarter note

double flats — lowers the pitch a whole step

double sharps — raises the pitch a whole step

duration — the distance in time of a musical event

dynamic marks — indicate the varying degree of loudness or softness at which the composition is to be played

event — in the Music Editor, an event is any single musical occurrence or symbol

flats — lowers the pitch one half step

f, ff, fff — forte, fortissimo, fortississimo; in sequence, loud, very loud and very very loud

H, I, J, K, L

harmonics — when a note is played, what is heard is the fundamental as well as the overtones or harmonics (which is an octave higher than the fundamental tone)

key signature — indicates the key the composition is in, also indicates which notes will be "sharped" or "flatted"

larghetto — tempo indicating fairly slow and broad

largo — tempo indicating very slow and broad

Ledger — (or leger) lines added above or below the staff to allow for notes which are too high or too low to be written on the five staff lines

lentissimo — tempo indicating fairly slow

lento — tempo indicating slow

letter notation — the use of letters to indicate note tones

M, N, O

measures — the space between two bar lines

meter — the number of beats per measure; e.g., if there are three beats per measure it is triple meter

mode — any seven pitch scale that does not follow the intervallic structure of the major or minor scales is a mode

moderato — tempo indicating medium speed

mf — mezzo forte or medium loud

mp — messo piano or medium soft

naturals — used to change a "sharpened" or "flatted" note to its "natural" scale

octave signs — indicates that something is either an octave lower or higher

P, Q, R

part — one of two or more voices in a multi-voice composition

pitch — the height or depth of a note

poly-rhythm — two or more strikingly contrasted rhythms occurring simultaneously

presto — tempo indicating very fast

proportional notation — the system of musical relationships of note values, rest values, etc.

rests — the periods of silence in music. Every note duration has an equivalent rest duration.

rhythm — regular occurrence of grouped strong and weak beats, or heavily and lightly accented tones, in alternation

S - Z

sharps — raises a note a half step

staccatissimo — in a staccato manner

staccato — means "detached". The notes are short and disconnected; indicated by a dot above or below them

staff — a series of five lines and four spaces on which symbols are written to indicate musical concepts

stars — (see asterisks)

stems — the thin vertical line which is drawn on the head of a note

sf, sfz — sforzando, indicates that a note be hit forcefully upon attack and then pulled back

tempo — the speed at which a piece is played

tenuto — from the Italian for "held, sustained"; to hold the note for its full value, and sometimes a bit longer

ties — joins two notes of the same pitch so it sounds like one continuous note

time — a term used loosely to indicate meter, tempo or the duration of a given note

vivace — tempo indicating lively

Appendix B
Music Editor Keyboard Commands

EDITOR CONTROL COMMANDS

Command	Function
ADDP	Add a part (name or number)
CHORDS	Set Editor to Chords mode
DELP	Delete a Part
LOAD	Load an existing file
NEW	Clear the Music Editor buffer
NOTES	Set Editor to Notes Mode
QUIT	Quit the Music Editor
SAVE	Save file in buffer to disk

CURSOR CONTROL COMMANDS

Command	Function
CTRL-A	Move cursor right
CTRL-Q	Move cursor left
CTRL-S	Scroll parts down
CTRL-W	Scroll parts up
GOTO n	Go to measure n (0-255)
GOTO 0	Go to beginning of part
GOTO 255	Go to end of part
>	Forward Delete
<	Backward Delete

SOUND CONTROL COMMANDS

Command	Function
FF	Absolute Dynamic, value = 100
FFF	Absolute Dynamic, value = 110
FT	Absolute Dynamic, value = 90
MF	Absolute Dynamic, value = 80
MP	Absolute Dynamic, value = 70
P	Absolute Dynamic, value = 60
PP	Absolute Dynamic, value = 50
PPP	Absolute Dynamic, value = 40
TEMPO n	Set Absolute Tempo (040-200)
TEMPO + (-) n	Set Relative Tempo (+/- 160)

NOTE MODIFIER COMMANDS

Command	Function
DYN n	Set Absolute Dynamic
DYN + (-) n	Set Relative Dynamic
FFFZ	Dynamic Accent
FFZ	Dynamic Accent
FZ	Dynamic Accent
NOAC	Remove Accent from Note or Chord
SF	Dynamic Accent
SFF	Dynamic Accent
SFFF	Dynamic Accent
SFFFZ	Dynamic Accent
SFFZ	Dynamic Accent
SFZ	Dynamic Accent
TIE	Tie two notes/chords together
!	Staccato accent
!!	Staccatissimo accent
^	Low percussive accent
^^	High percussive accent
%	Tenuto accent

SIGNATURE COMMANDS

Command	Function
ALTO	Alto Clef
BASS	Bass Clef
KEY 0	No Flats, No Sharps
KEY n # (*)	Set n sharps (flats)
SYSTEM	System Clef (Treble and Bass)
TENOR	Tenor Clef
TIME nn/mm	Set Time signature nn/mm nn = 2 through 32 mm = 1,2,4,8,16 or 32
TREBLE	Treble Clef

ACCIDENTALIZE COMMANDS

Command	Function
N	Naturalize this note
#	Sharp this note
##	Double Sharp this note
*	Flat this note
**	Double Flat this note

DURATION COMMANDS

Command	Function
EI	Set duration to Eighth note
H	Set duration to Half note
Q	Set duration to Quarter note
S	Set duration to Sixteenth note
T	Set duration to Thirty-second note
W	Set duration to Whole note
.	Add Dot to current duration

ASSORTED COMMANDS

Command	Function
CD	Change Duration of this note
CP	Change Pitch of this note
R	Insert Rest of current duration
SYNC n	Insert Sync mark number n
: n	Set Octave to n
A - G	Pitch selection
LOC n	Spatial Location, n = 0 - 9
LOC a	Spatial Location, a = LEFT, RIGHT or BOTH
INST a	Instrument Assignment, a = IDEF name; no spaces or key words allowed, i.e. BASS
PRINT	initiate Print Option

Appendix B - 2

Appendix C Error Messages

The MusicSystem produces several types of error messages; some are purely informational, some warn you of errors in the data the system is receiving, and some notify you that a fatal error has occurred. These messages are listed in this appendix along with an explanation of the message and suggestions for correction of the error.

Error messages common to all four MusicSystem programs are listed first, followed by Music Editor, Music Merger, Music Player and Instrument Definer messages. Finally, a set of fatal errors are listed; in typical use of the MusicSystem, these fatal errors will never occur. They are listed here only as reference.

The messages are listed alphabetically within each program. The first line on each page indicates which MusicSystem program generates the message.

* * *

Common Error Messages

The following error messages can occur during use of *any* of the MusicSystem programs, some are preceeded by "DISK ERROR:".

BAD SLOT OR DRIVE ENTRY

Meaning: Entry was not standard DOS syntax for slot, drive designation

Action: Try again.

CATALOG FULL

Meaning: There is no more room in the Catalog Directory.

Action: Command is cancelled. Processing continues.

DEVICE?

Meaning: Problem with disk, or disk drive.

Action: Command is cancelled.

DISK FULL

Meaning: The disk does not contain enough unused space to hold another file. Delete the file being SAVEd as it may have bad or incomplete data.

Action: Command is cancelled.

DISK WRITE PROTECTED

Meaning: This disk is write protected; you cannot save a file to it. Remove write protect sticker and repeat.

Action: Command is cancelled. Processing continues.

FILE DOES NOT EXIST

Meaning: The file named either does not exist or is not on the disk that is being searched.

Action: Command is cancelled. Processing continues.

FILE LOCKED

Meaning: File you are attempting to write to is locked.

Action: Command is cancelled. Processing continues.

FILE TYPE MISMATCH

Meaning: Incompatible file types, file created on earlier version of MusicSystem.

Action: Command is cancelled. Processing continues.

OUT OF MEMORY

Meaning: Insufficient memory for disk buffers, Apple memory may be suspect.

Action: Prompts "PRESS SPACE BAR WHEN READY", program is cancelled, returns to Main Menu.

VOLUME # MISMATCH

Meaning: Serious MusicSystem software problem. Save the file being edited to a new, initialized disk.

Action: System may abort; if so, returns to Main Menu.

SYSTEM MEMORY ERROR

Meaning: Space allocation problem, Apple memory may be suspect.

Action: Prompts "PRESS SPACE BAR WHEN READY", program is cancelled, returns to Main Menu.

* * *

Music Editor Error Messages

The following messages are only generated while using the Music Editor program.

ALREADY DOTTED

Meaning: Attempt to add a dot option to a duration that is already dotted.

Action: Command is cancelled. Processing continues.

BAD ACCIDENTAL

Meaning: Accidental in a KEY command must be "#" or "*".

Action: Command is cancelled. Processing continues.

BAD DYNAMIC
Meaning: Value entered for dynamic in DYN command outside the allowable range; range is -127 to +127.
Action: Command is cancelled. Processing continues.

BAD OCTAVE
Meaning: Value of octave entered is not allowed; octave must be from 0 to 7.
Action: Command is cancelled. Processing continues.

BAD NSF
Meaning: Attempt to enter more accidentals in a KEY command than is allowed, range of accidentals is 1 to 7.
Action: Command is cancelled. Processing continues.

BAD QPM
Meaning: Value entered in TEMPO command out of allowable range, range is 40 to 200 for absolute tempo, -160 to +160 for relative tempo
Action: Command is cancelled. Processing continues.

BAD SLOC
Meaning: Attempt to enter spatial location out of allowable range in LOC command, range is 0 to 9
Action: Command is cancelled. Processing continues.

CAN'T FIND PITCH
Meaning: During Change Pitch command, the "target" pitch cannot be found.
Action: Command is cancelled. Processing continues.

DUPLICATE PITCH
Meaning: Occurs in chords mode, when a note is input two times
Action: Press space bar and continue

ILLEGAL SYMBOL
Meaning: Illegal keyboard symbols: $, &, ', (,), =, @, ;, or ?
Action: Command is cancelled. Processing continues.

ILLEGAL TIMESIG
Meaning: Value used in TIME command is not allowed; see TIME command for range of allowable values.
Action: Command is cancelled. Processing continues.

? INTEGER AFTER +/-
Meaning: No integer is found after a + or - symbol in a command.
Action: Command is cancelled. Processing continues.

INTEGER > 255 NOT ALLOWED
Meaning: No integer greater than 255 may be used in Music Editor.
Action: Command is cancelled. Processing continues.

ITEM IS NOT A CHORD
Meaning: A chord-type command was applied to a musical event that was not a note or chord.
Action: Command is cancelled. Processing continues.

MAX POLYPHONY EXCEEDED
Meaning: Chords must not have more than 15 notes per chord.
Action: Command is cancelled. Processing continues.

MUST BE PART 1
Meaning: Attempt to enter TEMPO in a part other than PART 1.
Action: Command is cancelled. Processing continues.

NO SCORE PRINTED
Meaning: Print command used, nothing in memory to print
Action: Press space bar to continue

OUT OF BUFFER SPACE IN INPUT FILE
Meaning: Attempt to load a "merged" file in a LOAD command.
Action: Command is cancelled. Reboots to Main Menu.

OUT OF SPACE
Meaning: No room for more insertion of musical events in the Music Editor buffer. Save file, or remove data from file to allow for more insertions.
Action: Command is cancelled. Processing continues.

PART NOT FOUND
Meaning: Part name or number in DELP command is not found.
Action: Command is cancelled. Processing continues.

SYNTAX ERROR
Meaning: Any command entered incorrectly.
Action: Command is cancelled. Processing continues.

TOO MANY PARTS
Meaning: Attempt to add more than 16 parts in one composition.
Action: Command is cancelled. Processing continues.

Bell on Apple speaker sounds
Meaning: Cursor at end of file or beyond parts; illegal CTRL-char.
Action: Command is cancelled. Processing continues.

* * *

Music Merger Error Messages

The following error messages may occur while using the Music Merger program. These messages may be preceeded by one of: "DISK ERROR:", "ERROR:", or "**ERROR**".

COMP FILE CAN'T = MERGE FILE
Meaning: The name specified for the MERGE file cannot be the same as one of the COMP files being merged.
Action: Prompts "PRESS SPACE BAR WHEN READY", returns to first Merger prompt.

NOT ENOUGH DATA WRITTEN TO FILE
Meaning: Disk is full and has no room for merged file.
Action: Prompts "PRESS SPACE BAR WHEN READY". Delete bad MERGE file, re-Merge on disk that is not full

OUT OF BUFFER MEMORY
Meaning: More data in the two MERGE files than can fit into Merge Buffer at one time.
Action: Prompts "PRESS SPACE BAR WHEN READY". Music Merger cancelled, returns to Main Menu. Delete bad MERGE file; make smaller files and re-try MERGE.

* * *

Music Player Error Messages

The following error messages may occur while using the Music Player programs, some are preceeded by either "— | ERROR :" or "*** ERROR **"

BAD FILE — RAN OUT OF DATA
Meaning: File contained insufficient data; i.e., disk became full before file was completely saved from Merger
Action: This file cannot be played.

BAD COMPOSITION FILE
Meaning: Bad disk; possible machine/software problem
Action: Prompts "PRESS SPACE BAR WHEN READY". Processing continues.

BAD. ITEM. TYPE
Meaning: Bad disk, no user recovery possbile.
Action: Prompts "PRESS SPACE BAR WHEN READY", returns to Main Menu

BAD. OV. STATE
Meaning: Bad disk, no user recovery possible.
Action: Prompts "PRESS SPACE BAR WHEN READY", returns to Main Menu

BAD PART NUMBER
Meaning: Response to "PART # TO CHANGE" was not within range
Action: Re-try using correct response.

BAD. PITCH. COUNT
Meaning: Bad disk, no user recovery possible.
Action: Prompts "PRESS SPACE BAR WHEN READY", returns to Main Menu

BAD PLAYER FILE
Meaning: 1. Bad disk or possible machine/software problem.
2. May be incomplete file; i.e. disk became full before file was completely saved from Player.
Action: Prompts "PRESS SPACE BAR WHEN READY". Re-compile and SAVE this composition from COMP. file.

BIN.PLAYER FILE IS MISSING
Meaning: Bad disk; BIN.PLAYER file missing from System disk.
Action: Prompts "PRESS SPACE BAR WHEN READY", SAVE the composition, use new copy of System disk.

INVALID TEMPO ALIGNMENT
Meaning: Tempo markers must be on note boundary.
Action: Music Player cancelled. See TEMPO command in Music Editor reference chapter.

MUSIC BOARD NOT IN MACHINE
Meaning: Self-explanatory
Action: Prompts "PRESS SPACE BAR WHEN READY", SAVE file, then install MusicSystem boards.

NO PLAY OR COMP FILE
Meaning: PLAY or COMP file named does not exist.
Action: Prompts "PRESS SPACE BAR WHEN READY". Check for file name spelling, or correct disk.

NO ROOM FOR INSTRUMENT DEFINITIONS
Meaning: 1. Too many unique instruments. 2. Play file too big to allow bonding of instrument.
Action: Prompts "PRESS SPACE BAR WHEN READY". 1. Use simpler instruments (fewer oscillators), or reduce the number of unique instruments.
2. Reduce size of Comp file and re-process in Player.

OBJECT CODE MEMORY OVERFLOW
Meaning: Insufficient room to create tables of data required by the PLAY file being created during a SAVE.
Action: Prompts "PRESS SPACE BAR WHEN READY". Song can be played, but you must remove some music if you want to SAVE it. Or set all instruments to same.

OUT OF FREE SPACE
Meaning: File is too large to be played.
Action: Music Player cancelled. Reduce size of each component section, re-MERGE, re-process.

OUT OF LEFT OSCILLATORS
Meaning: Too many oscillators used on Left channel.
Action: Prompts "PRESS SPACE BAR WHEN READY". Re-assign some to RIGHT, or use simpler instruments.

OUT OF LOGICAL OSCILLATORS
Meaning: Combination of instruments is too complex; requires more oscillators than are available.
Action: Prompts "PRESS SPACE BAR WHEN READY". Use simpler or fewer instruments

OUT OF PHYSICAL OSCILLATORS
Meaning: Instrument assignments inconsistent with available physical oscillators. Reduce or regroup assignments.
Action: Prompts "PRESS SPACE BAR WHEN READY". Change any BOTH to LEFT or RIGHT assignments. Or change all to MONO or simpler instruments.

OUT OF RIGHT OSCILLATORS
Meaning: Too many oscillators used on Right channel.
Action: Prompts "PRESS SPACE BAR WHEN READY". Re-assign some to LEFT, or use simpler instruments.

TEMPO. VOICE
Meaning: Bad disk, no user recovery possible.
Action: Prompts "PRESS SPACE BAR WHEN READY", returns to Main Menu

TOO MANY PARTS IN COMPOSITION
Meaning: Bad disk; possible machine/software problem
Action: Prompts "PRESS ANY KEY TO CONTINUE". Processing continues.

TOO MUCH POLYPHONY IN COMPOSITION
Meaning: If total of all max polyphony sizes is greater than 16, this composition cannot be played.
Action: Prompts "PRESS SPACE BAR WHEN READY". Make the chords smaller or delete PARTS.

* * *

Instrument Definer Error Messages

The following error messages may occur while using the Instrument Definer program, some are preceeded by "ERROR:".

BAD COORDINATE ENTRY

Meaning: Coordinates must be entered in the form : X,Y

Action: Press Space Bar, try again.

EVEN NO. EXPECTED

Meaning: For "Select Attack" and "Select Decay" times, even number is expected.

Action: Press Space Bar, select even number in range.

IDEF DOES NOT EXIST

Meaning: IDEF cannot be saved it it is not in memory.

Action: Press Space Bar, select another IDEF option.

INCOMPLETE IDEF FILE WRITTEN

Meaning: Possibly ran out of diskette space.

Action: Re-save onto another diskette.

NO FREQUENCY POINTS

Meaning: Instrument definition is not complete.

Action: Define frequency points with Coordinates Editor

NO LOGICAL OSCILLATORS

Meaning: Logical oscillators must exist before creating waveform and before displaying instrument information.

Action: Press space bar, add logical oscillator first.

NUMBER RANGE IS X TO Y

Meaning: The value was not within range X to Y

Action: Press space bar, try again.

OUT OF HEAP SPACE SAVE. IDEF

Meaning: Not enough stack space; program may be damaged.

Action: Re-copy program from master disk; re-try.

OUT OF HEAP SPACE: LOADING

Meaning: Not enough stack space; program may be damaged.

Action: Re-copy program from master disk; re-try.

OUT OF HEAP SPACE: SAVE WAVEFORM

Meaning: Not enough stack space; program may be damaged.

Action: Re-copy program from master disk; re-try.

OUT OF LEFT PHYSICAL OSCILLATORS

Meaning: Too many oscillators used on Left channel

Action: Prompts "PRESS SPACE BAR WHEN READY". Re-assign some to RIGHT, or use simpler instrument

OUT OF LOGICAL OSCILLATORS

Meaning: Combination of instruments is too complex; requires more oscillators than are available.

Action: Prompts "PRESS SPACE BAR WHEN READY". Use simpler or fewer instruments

OUT OF RIGHT PHYSICAL OSCILLATORS

Meaning: Too many oscillators used on Right channel

Action: Prompts "PRESS SPACE BAR WHEN READY". Re-assign some to LEFT, or use simpler instrument

OSCILLATOR DEFINITION INCOMPLETE

Meaning: From Add Logical Oscillators; logical oscillator must have wavename, osc weight, exp. decay factor, amplitude history in order to be minimally complete.

Action: Press space bar, complete the oscillator definition

TOO MANY AMPLITUDE PAIRS GENERATED

Meaning: More than 15 pairs generated

Action: Delete other pairs to input this pair

TOO MANY LOGICAL OSCILLATORS

Meaning: The IDEF already has eight logical oscillators

Action: Press space bar, pick another IDEF option

TROUBLE LOADING IDEF FILE

Meaning: Diskette may possibly be bad

Action: Press space bar when ready

X COORDS MUST BE IN ASCENDING ORDER

Meaning: The X coordinates must be in ascending order

Action: Press space bar, try again.

Fatal errors in the Instrument Definer should not occur during normal use of the Music-System. These messages are listed here for reference purposes only. All such messages appear like this:

FATAL ERROR # n

The meaning of the number n displayed in the message follows:

#1 — Out of Heap Space
#2 — IDEFDOS.OVLY—the overlay file was not found when needed
#3 — Trouble reading IDEFDOS.OVLY (not all sectors were read)
#4 — System memory problem (heap space returned out of order)
#5 — BIN.IDEF not found during IDEF initialization
#6 — Trouble reading BIN.IDEF
#7 — HIRESTABLES not found
#8 — Trouble reading HIRESTABLES

* * *

Fatal Error Messages

The following error messages should never occur during normal use of the MusicSystem. If they do occur, be sure to write down the error message before continuing. These messages indicate that something extremely serious has occurred with the MusicSystem software. Do not continue to use a copy of the software that produces any of these errors; rather, make a new copy from the master disk and use it. Some of the messages are preceeded by "DISK ERROR:".

CAN'T READ
Meaning: Indicates bad disk.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

IN SOFTWARE—CALL MHI
Meaning: Possible machine failure or software problem
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

MACRO STACK OVERFLOW
Meaning: Bad disk; possible machine/software problem.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

OUT OF HEAP SPACE
Meaning: Possible machine failure
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

OUT OF STACK SPACE
Meaning: File I/O problem.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

OV HEAP SPACE
meaning: Bad disk; possible machine/software problem.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

OV OPEN
Meaning: System disk is possibly bad, or in wrong drive.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

OVLY.LOAD FAILED
Meaning: System disk is possibly bad.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

SPACE RELEASE PROBLEM
Meaning: Bad disk; possible machine/software problem.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

SYSTEM MEMORY PROBLEM
Meaning: Possible machine failure
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

TABLE DATA NOT FOUND
Meaning: Utility Tables "TABLE1" and/or "TABLE2" not on disk.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

TROUBLE READING TABLE DATA FILE
Meaning: Possible bad disk.
Action: Record Error Message, SAVE file if possible to new disk. Re-boot and try again.

* * *

Appendix D MusicSystem Files

The Mountain Computer MusicSystem uses several different types of files. The Music Editor accepts input from the user, displays or prints it as a musical score and stores it as COMP (composition) files. The Instrument Definer accepts input from the user, reflects it as statistics or audio feedback and stores it as IDEF (instrument definition) and WAVE (waveform) files. The Music Player converts COMP files to an internal form, bonds instrument definitions to the data and stores them as PLAY (playable) files. When you CATALOG your song and instrument file diskettes, you will see files with the prefixes COMP., IDEF., WAVE., AND PLAY.

File Maintenance
Depending on your particular use of the MusicSystem, you will want to devise a file storage system to allow the most convenient access to your data. For example, some users may want to group input COMP files, merged COMP files, IDEF and WAVE files and the resultant PLAY file on one diskette for each composition. Other users will find it helpful to keep all file types on separate diskettes and label them appropriately; i.e., IDEF #30, PLAY #4, etc. Naming files with descriptive and version identifying names will also help.

Moving Files from Diskette to Diskette
If you wish to move files from one diskette to another, you may use either of two methods.

1. Using the existing MusicSystem programs to copy files from one diskette to another. The program you use depends on the type of file you want to move.

File Type	Program	Steps
PLAY	Music Player	1. Load the PLAY filename 2. Choose "4" from Menu 3. Change diskette to SAVE file to desired diskette.
IDEF	Instrument Definer	1. Load the instrument definition w/choice "1" 2. Choose "2" from Menu 3. Change diskette to SAVE file to desired diskette.
WAVE	Instrument Definer	1. Load the instrument definition w/choice "1" 2. Choose "6" from Menu 3. Choose SAVE option ("5"). 4. Change diskette to SAVE file to desired diskette.
COMP	Music Merger	1. Load the COMP filename 2. Hit RETURN at prompt for 2nd file name 3. Change diskette to COPY file to desired diskette.

2. The other method to move files can only be used on PLAY, IDEF and WAVE files. Begin at the Main Menu and type

```
BLOAD PLAY.filename
```

Change the diskette to the desired storage diskette and type

```
BSAVE PLAY.filename, A$6600, L$yyy
```

In this BSAVE, $6600 is the BLOAD address and yyy is the BLENGTH.

The same method may be used for all IDEF and WAVE files, however the BLOAD address is $8800.

This chapter provides details of file formats for all types of files in the MusicSystem. A CATALOG program such as the one available on Mountain Computer's COPYROM (part # MHP-X026) is an easy way to find the value of the length parameters.

File Formats: Version 2.0
The following pages provide format specifications for the four types of MusicSystem file types: COMP, PLAY, IDEF and WAVE. This information is not necessary for the regular use of the MusicSystem, but may be helpful to some programmers.

* * *

COMP File Format Specifications
A COMP file includes a header section and a composition data section. The length of the header section depends upon the number of parts in the composition. The length of the data section depends upon the length of the song.

COMP File Header Section Format

# of bytes		Item	Description
1		NV	number of input parts
1	11	MPS_1	maximum polyphony size for part 1
1		BYT_1	relative byte location, part 1
1		SEC_1	relative sector location, part 1
8		S_1	part 0 name, ASCI I

.
.

more part header data, 11 bytes per part

.
.

# of bytes		Item	Description
1	11	MPS_{nv}	number of input parts, last part
1		BYT_{nv}	number of input parts, last part
1		SEC_{nv}	relative sector loc., last part
8		S_{nv}	last part name, ASCII

The Composition Data Section Format
The format for the composition's data section is a stream of input items for each part with an "end-of-voice" marker after the items associated with each voice. Each input item is preceeded by a single byte type of item indicator; the length of each data item is variable. Each input item looks like this:

# of bytes	Item	Description
1	Type	type of item
L	data	data input information

A chart of descriptions and length in bytes of each of the possible types of data for COMP files follows.

Type #	Item	Length in bytes
0	end of voice	0
1	chord	3 + pitch count
2	rest	1
3	whole rest	2
4	clef	1
5	time signature	2
6	key signature	1
7	undefined	0
8	undefined	0
9	undefined	0
10	dynamic	2
11	tempo	2
12	spatial location	1
13	instrument name	11
14	undefined	0
15	undefined	0
16	measure bar	0
17	start of part	0

* * *

PLAY File Format Specifications
A PLAY file includes a header, a song file and a trailer.

PLAY File Header Section Format

# of bytes		Item	Description
1	2	low byte	BLOAD address ($6600)
1		high byte	
1	2	low byte	BLENGTH
1		high byte	
1		NV	number of input voices
1		V	version number
1	2	low byte	trailer offset with respect to BLOAD address
1		high byte	
1	12	NO_1	number of voices for part1
11		VN_1	part name

more part header data, 12 bytes per part

# of bytes	Field	Description
1	NO_i	number of voices for part i
11	VN_i	part name for part i

12

# of bytes	Field	Description
1	NO_{nv}	number of voices for last part
11	VN_{nv}	part name for last part

12

PLAY FIle Data Section Format

The format of the data section for PLAY files is a stream of output items with the following format:

# of bytes	Item	Description
1/2 (4 bits)	ID	type of item
1/2 (4 bits)	v	voice number (0-15)
L	data	data output information

A chart of descriptions and length in bytes of each of the possible types of data for PLAY files follows.

ID #	Item	Length in bytes
0	undefined	0
1	undefined	0
2	undefined	0
3	undefined	0
4	tempo change	1 implicit rest, # of interrupts
5	global dynamic change	1 8 bit 2's complement
6	local dynamic change	1 8 bit 2's complement
7	undefined	0
8	extended rest	2 # of whole rests
9	regular rest	see Note 1
10	notes	see Note 2

Note 1: The data portion of a regular rest is described by either one of two possible duration formats. The format for a 1-byte duration is:

# of bits	Value	Description
1	0	indicates 1 byte duration
1	0	indicates ornamental flag off
6		describe # of interrupts

The format for a 2 byte duration is:

# of bits	Value	Description
1	1	indicates 2 byte duration
1	0	indicates ornamental flag off
6		hi order value interrupt count
8		low order value interrupt count

Note 2: The data portion of a note consists of three sections: duration, ornament byte and pitch. The duration data formats are described above; the ornament flag may be on ("1").

The format of the ornament byte is:

# of bits	Field	Description
2	ST	staccato flag; 1 = low, 2 = high
2	Pe	percussive flag; 1 = low, 2 = high
1	Te	tenuto flag; 1 = on
1	Ti	tie flag; 1 = on
1		undefined
1		undefined

The pitch data is described by one of two formats.

The format for a 1-byte pitch is:

# of bits	Value	Description
1	0	indicates 1-byte pitch format
7		pitch table offset

The format for a 2-byte pitch is:

# of bits		
1	1	indicates 2-byte pitch format
7		high order value of absolute pitch value
8		low order value of absolute pitch value

In summary, the complete format of a note item in a song file is:

# of bytes	Description	
1/2 (4 bits)	ID	type of item
1/2 (4 bits)	V	voice number
1 or 2	DURATION	either 1 or 2 bytes
1	ORNAMENT	1 = on, 0 = off
1 or 2	PITCH	either 1 or 2 bytes

The format of the TRAILER is:

# of bytes		item	Description
1	2·NV	S_1	speaker assignment, part 1
1		I_1	instrument number, part 1
.			
.			
1		S_i	speaker assignment, part i
1		I_i	instrument number, part i
.			
.			
1		S_{nv}	speaker assignment, last part
1		I_{nv}	instrument number, last part
1	(11·NI)+1	NI	number of unique instruments
11		N_1	name of instrument 1
.			
.			
11		N_{NI}	name of last instrument
2	2·NV	$IDEFDTR_1$	location of instrument definition for input part 1
.			
.			
2		$IDEFDTR_{nv}$	location of instrument definition for last input part

The instrument definitions are loaded directly after the PLAY file trailer. The waveforms (WAVE files) are loaded at the first page boundary after the last IDEF file.

The format of an IDEF file is:

# of bytes	Item	Description
2	low byte high byte	BLOAD address ($8800)
2	low byte high byte	BLENGTH
1	N	number of logical oscillators
1	ATAKTIME	attack time expressed in interrupts
1	DECAYTIME	decay time expressed in interrupts
1	LOGLIN	linear flag ($80) or log flag ($0)
32	DECAY TABLE	decay slope table
2*N	SUS. DECAY	exponential decay factor (internal)
2*N	EXP DECAY	exponential decay factor (external)
1*N	WEIGHT	oscillator weights
11*N	WAVENAME	waveform table names
1	N_F	# of bytes for frequency pairs
N_F	FREQ. PTS	time ordered frequency history coordinates—see Detail 1
1	N_A	# of bytes for amplitude pairs
N_A	AMPL.PTS	time ordered amplitude history coordinates—see Detail 2
1	N_{FI}	# of input frequency pairs
N_{FI}	FREQ.PAIRS	input frequency pairs—see Detail 3
1	N_{AI}	# of input amplitude pairs
N_{AI}	AMPL.PAIRS	input amplitude pairs—see Detail 4

Detail 1: Each time-ordered frequency history pair is formatted:

# of bits	Item	Description
4	ΔT	# of interrupts (1 to 16)
4	ID	oscillator number
1	S	sign bit (1 = positive, 0 = negative)
7	F	frequency change (0 - 127)

Detail 2: Each time-ordered amplitude history pair is formatted:

# of bits	Item	Description
4	AF	positive fractional slope
4	ΔT	# of interrupts (1 to 16)
8	AW	whole slope (-127 to +127) (2's complement)

Detail 3: Each input frequency pair is formatted:

# of bits	Item	Description
1	T	time (1 - 255)
1	F	frequency (-127 to +127) (2's complement)

Detail 4: Each input amplitude pair is formatted:

# of bits	Item	Description
1	T	time (1 - 255)
1	FM	amplitude (0 to 127)

WAVE File Format Specifications

There are two types of wave files, ordinary waveforms and waveforms which have been generated by the Instrument Definer program WAVEMAKER. The MusicSystem can play instruments with either type. The formats for each are described here.

The format of an ordinary wave file is:

# of bytes	Description	
2	low byte high byte	BLOAD address ($8800)
2	low byte high byte	BLENGTH ($100)
256		waveform data

The format of a WAVEMAKER wave file is:

# of bytes	Description	
2	low byte high byte	BLOAD address ($8800)
2	low byte high byte	BLENGTH ($118)
256		waveform data
24		harmonic weights

Appendix E
Bibliography and Acknowledgements

Books

The most extensive bibliography of books dealing with computer music can be found in the pages of the Computer Music Journal. Contact them for current information regarding books of interest to owners of the MusicSystem.

Any standard musical theory and notation text can be of assistance to the computer scientist/hobbyist needing a source of information about conventional music. The developers of the MusicSystem based their musical research on:

Music Notation A Manual of Modern Practice by Gardner Read, 1969. Published by Allyn and Bacon, Inc. 470 Atlantic Avenue, Boston, Massachusetts.

A great book for beginning music students who need fundamental information about music notation and theory is: *Learning to Read Music* by Robert Lilienfeld, 1979. Published by Barnes and Noble Books, New York, New York.

Periodicals

Computer Music Journal
Box E
101 Doyle Street
Menlo park, CA 94025

BYTE—the small systems journal
170 Main Street
Peterborough, NH 03485

Interface Age
16704 Marquardt Ave.
Cerritos, Ca. 90701

Acknowledgements and Thanks

We would like to express our appreciation and thanks to the following individuals who contributed to the developemnt of the MusicSystem:

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Bob Hoover — provided assistance in the early stages of system and hardware design.

Margee Milks — provided editorial assistance and review of the Operating Manual.

Appendix F
Specifications of Sound Generation

The MusicSystem is a sound generator, as opposed to a sound reproducer. Thus, the specifications for performance of the MusicSystem are unique and should not be compared to sound reproduction systems. However, some figures can be discussed to explain the abilities of the MusicSystem in generating sound and music. In any case, the better your stereo system reproduces audio input, the better the final listening experience will be when using the MusicSystem.

1) Frequency output range is roughly 30 to 13000 Hz. Frequency output is a measurement of the range of audio frequencies that the MusicSystem can produce.

2) Sample rate is 32k Hz. Sample rate is the number of times per second that the waveform table can be read. The theoretical maximum frequency that can be synthesized by the MusicSystem is one half the sample rate or 16k Hz. However, this requires ideal filters to separate frequencies close to 16k Hz. As a result, the practical limitation of the board is 13K Hz.

3) Frequency resolution is 1/2 Hz. Frequency resolution determines the "size" of each frequency step that can be produced by the MusicSystem. Thus, the MusicSystem can produce a frequency step of 1/2 Hz or greater. All frequencies in the MusicSystem are integral multiples of 1/2 Hz steps.

4) Waveform resolution is 8 bits. Waveform resolution is the degree of precision that the waveform is stored in memory. Waveform resolution is 8 bits times 256 samples per waveform. because waveforms are stored as tables that are 256 bytes long.

5) The MusicSystem has two distinct volume determining functions, each with 256 levels of control. One is a volume control for each oscillator; the other is an overall volume setting for all the oscillators. The volume setting for each oscillator is used to provide a waveform envelope. The overall setting determines the setting of all 16 oscillators together.

The signal amplitude of any oscillator is a product of its amplitude in the waveform table times its own volume setting times the overall volume setting.

6) The MusicSystem hardware has been carefully tuned to a "stretch" scale which deviates slightly from the standard tempered scale. This tuning has been shown to produce the most accurate audio result from the board. The stretch scale is 8% lower than the tempered scale at the low end and 10% higher at the high end. Our tuning chart follows:

Pitch ID #	Frequency (Hz)	Pitch IS #	Frequency (Hz)
0	33	49	567
1	35	50	601
2	37	51	637
3	39	52	675
4	41	53	715
5	44	54	758
6	46	55	803
7	49	56	851
8	52	57	902
9	55	58	955
10	59	59	1012
11	62	60	1073
12	66	61	1137
13	70	62	1204
14	74	63	1276
15	79	64	1352
16	83	65	1433
17	88	66	1518
18	94	67	1608
19	99	68	1704
20	105	69	1806
21	112	70	1913
22	119	71	2027
23	126	72	2149
24	133	73	2277
25	141	74	2413
26	150	75	2558
27	159	76	2711
28	168	77	2873
29	178	78	3054
30	189	79	3227
31	200	80	3420
32	212	81	3624
33	225	82	3841
34	238	83	4071
35	252	84	4316
36	267	85	4577
37	283	86	4852
38	300	87	5145
39	318	88	5455
40	337	89	5874
41	357	90	6132
42	378	91	6502
43	401	92	6894
44	425	93	7309
45	450	94	7750
46	477	95	8217
47	505		
48	535		

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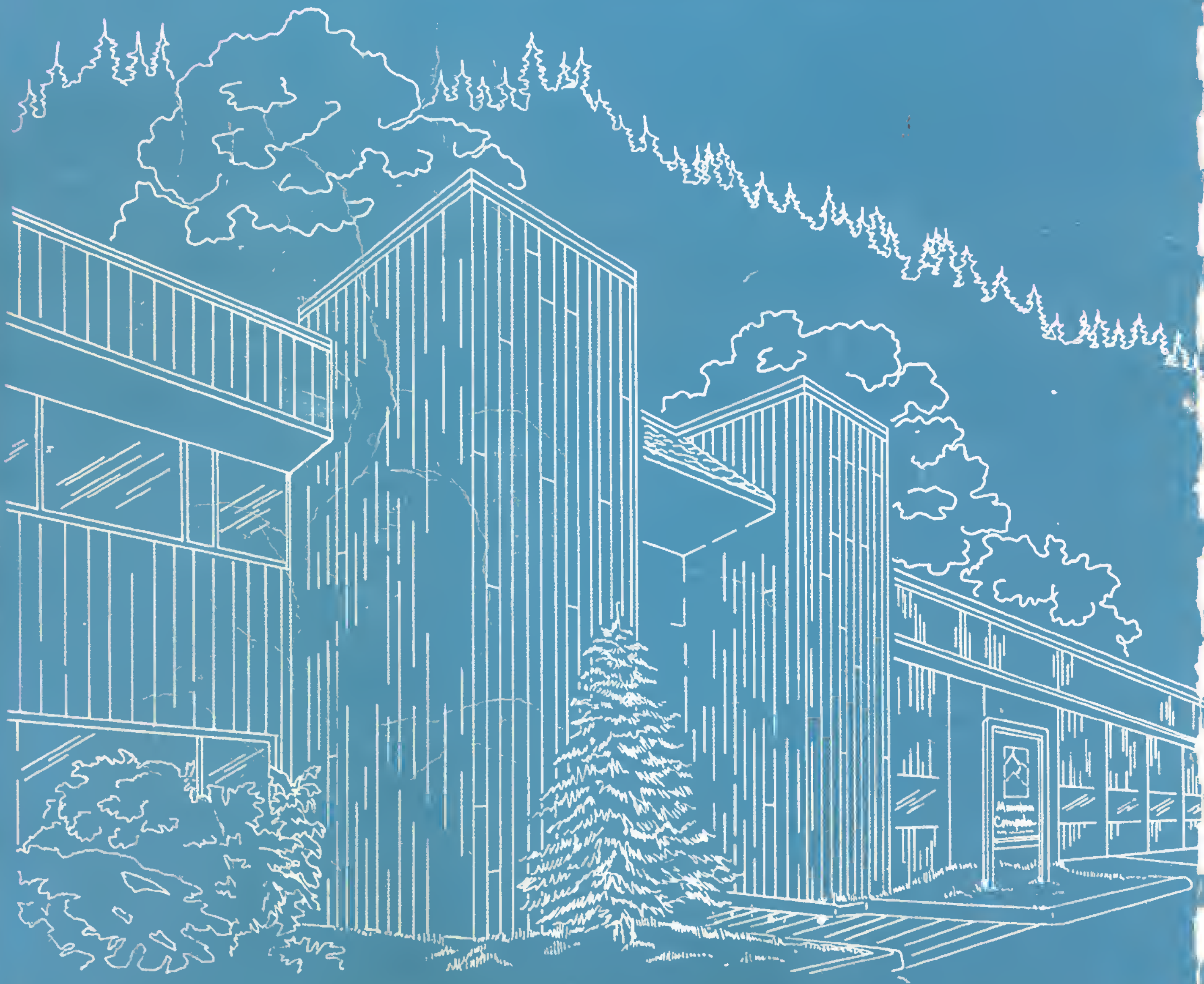

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